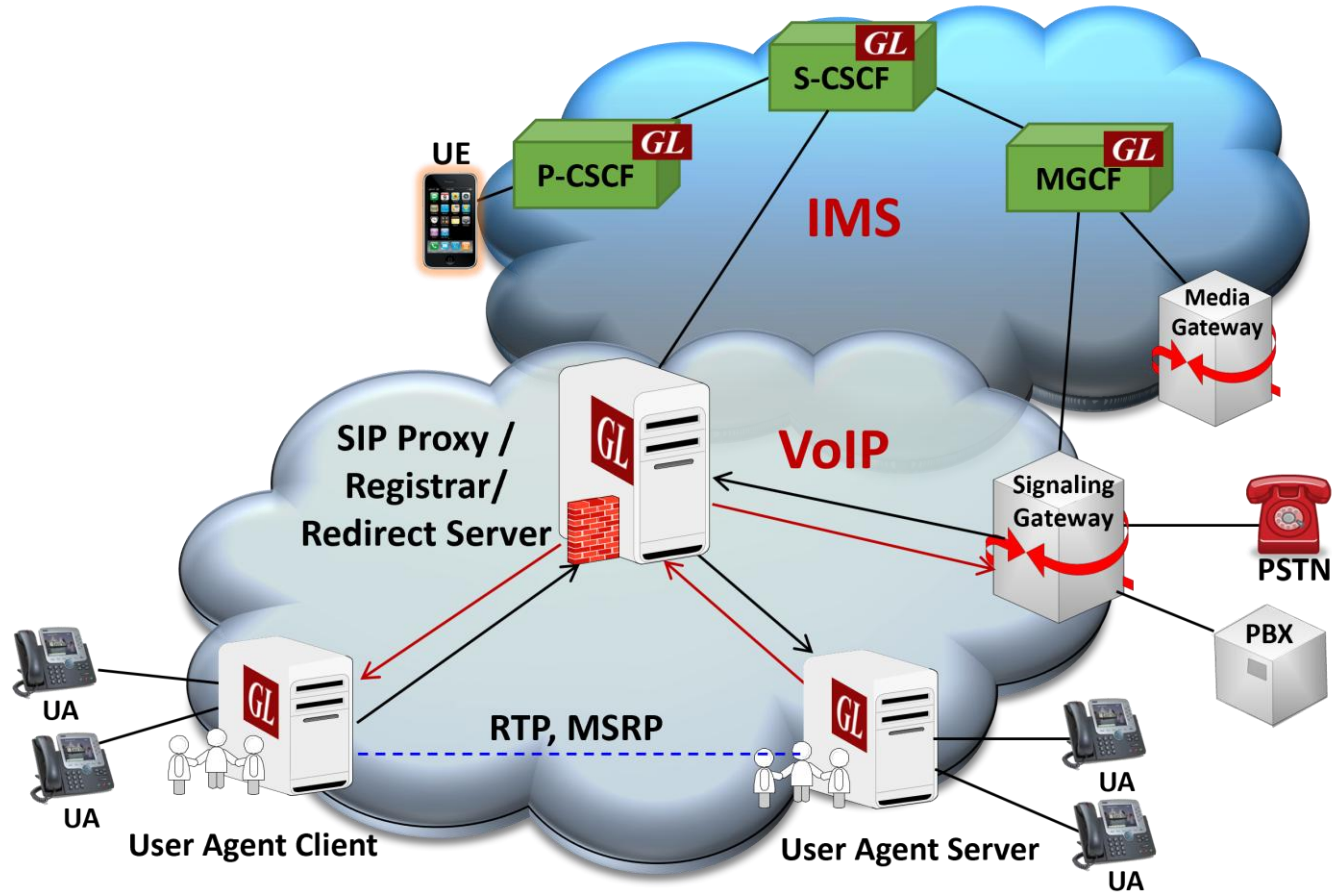

MAPS™ SIP SIP + RTP + MSRP Simulation

8 April 2026



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MAPS™ SIP

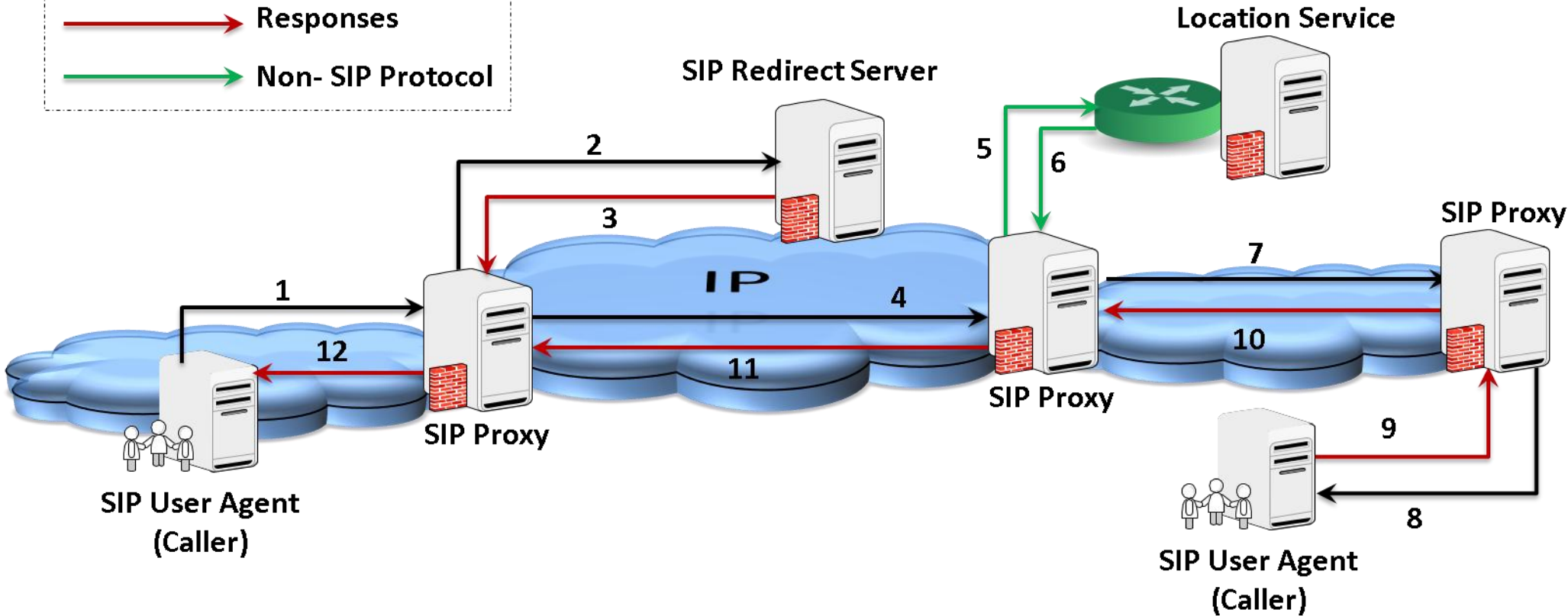
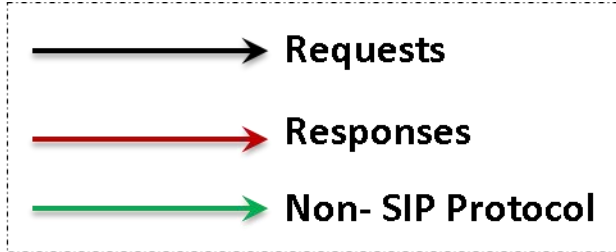


MAPS™ SIP with RTP Traffic Generation
 (2500 simultaneous calls)
MAPS™ SIP Conformance

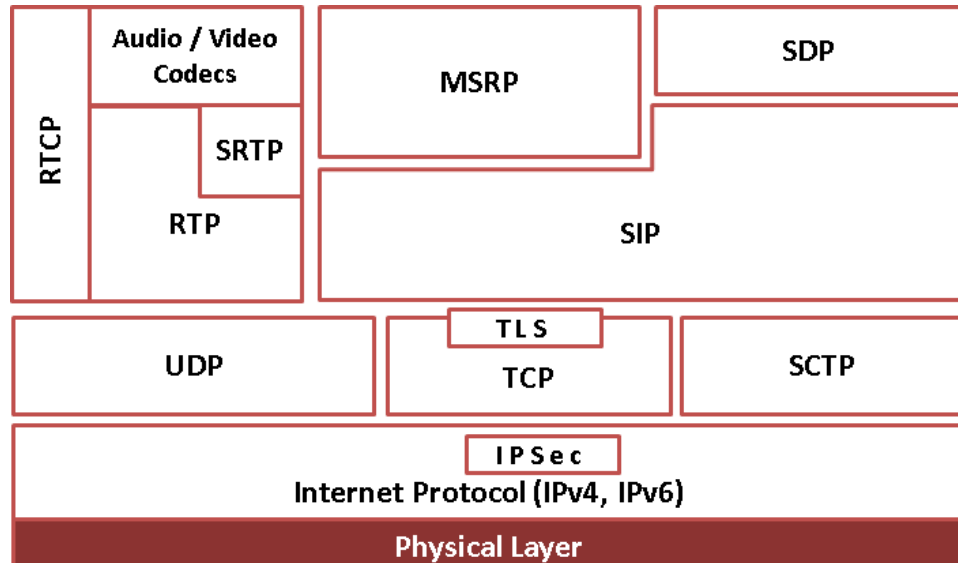


MAPS™ SIP (w/ 8 x 10 Gbps Ethernet Ports)
HD RTP Traffic Generator
 160,000 Simultaneous Calls (with RTP Traffic)

SIP Architecture and Entities

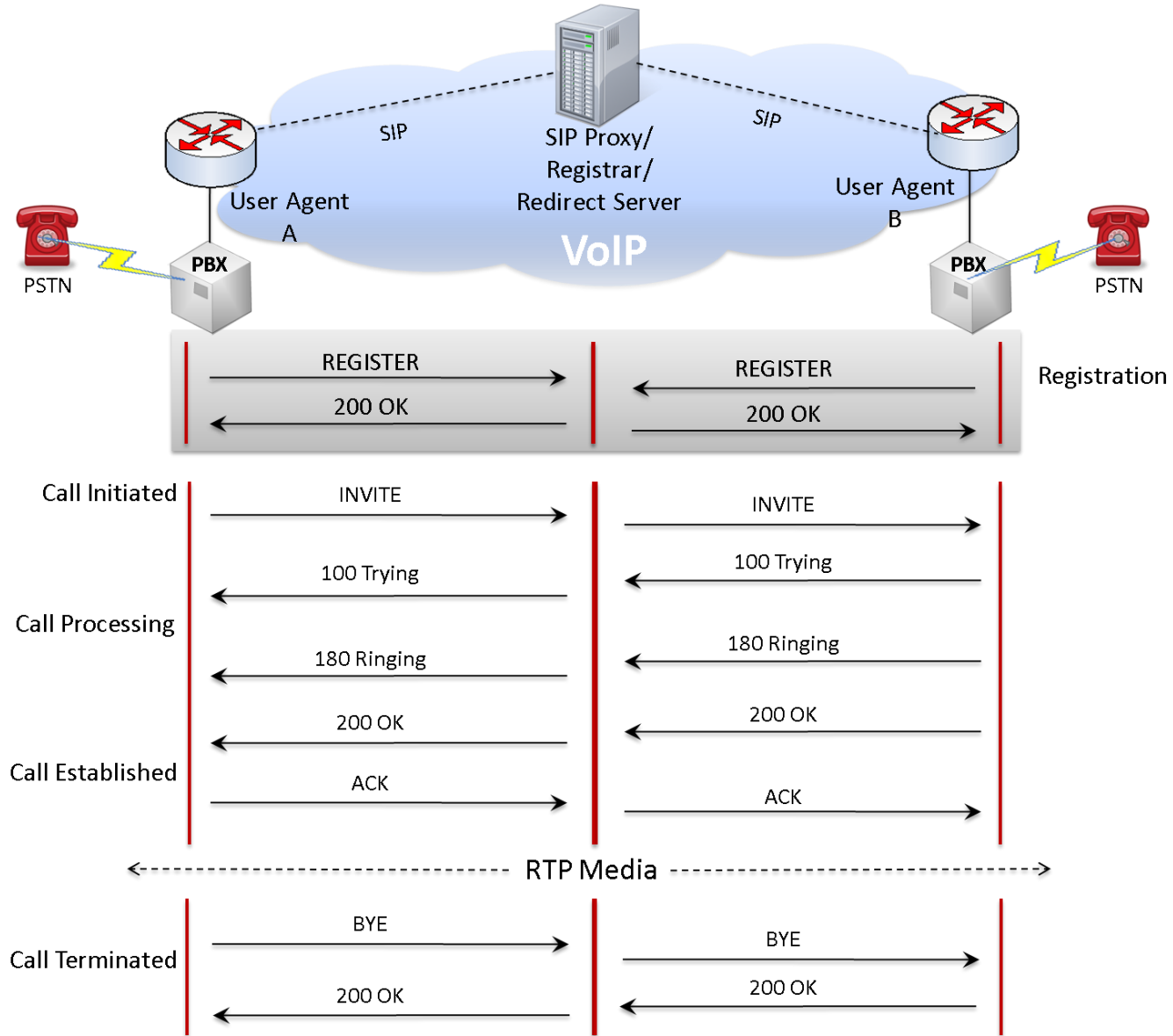


SIP Protocol Stack



Supported Protocols	Standard / Specification Used
SIP SIP Conformance	RFC 3261 ETSI TS 102-027-2 v4.1.1
SIP Extensions	RFC 3262 - Reliability of Provisional Responses in the Session Initiation Protocol (SIP) RFC 3311 - The Session Initiation Protocol (SIP) UPDATE Method RFC 3455 - Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP) RFC 3515 - The Session Initiation Protocol (SIP) Refer Method RFC 3310 - HTTP/SIP Digest Authentication Using Authentication and Key Agreement (AKA) RFC 3263 - Session Initiation Protocol (SIP): Locating SIP Servers
Secure Real-time Transport Protocol (SRTP)	RFC 3711 - Secure Real-time Transport Protocol (SRTP) RFC 3551 - Standard 65, RTP Profile for Audio and Video Conferences with Minimal Control)
Message session Relay Protocol (MSRP)	RFC 4975 - Message Session Relay Protocol (MSRP)

Generic SIP Call Flow



MAPS™ SIP Variants

MAPS™ SIP Software with Notebook PC

MAPS™ SIP Protocol Test Tool (Item # PKS120):

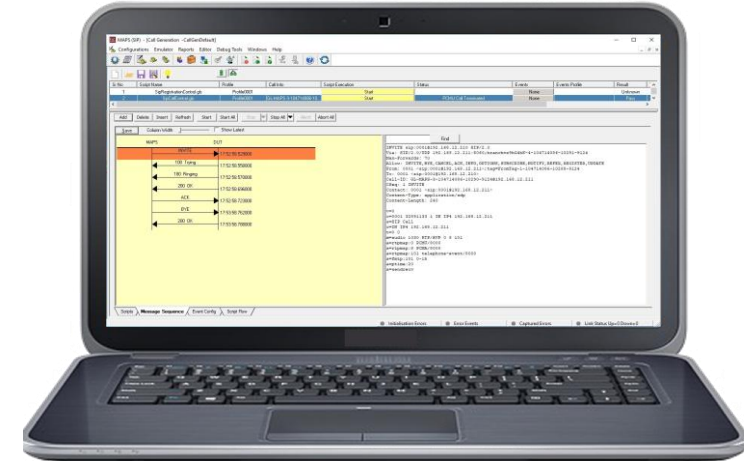
- RFC 3261 - Primary SIP standard
- RFC 3262 - PRACK
- RFC 3515 - REFER

MAPS™ SIP Conformance Suite (Item # PKS121):

- ETSI TS 102-027-2 v4.1.1 (2006-07) - 300+ scripts designed to test SIP UAs for conformance to RFC 3261

MAPS™ SIP High Density (HD) (Item # PKS109):

- MAPS™ SIP HD emulates up to 160,000 simultaneous calls using 8 Gigabit Ethernet ports
- Available in portable or rack-mount form factors



MAPS™ SIP HD



8x1/10 GigE

High Performance Smart NIC

MAPS™ SIP Highlights

signaling	<ul style="list-style-type: none">• Generates and processes SIP valid and invalid messages• Supports complete customization of SIP headers, call flow, and messages• Supports complete customization of scripts and parameters in the profiles• Each SIP message template facilitates customization of the protocol fields and access to the various protocol fields from the scripts• Supports IPv4 /IPv6 and transport over UDP and TCP, and TLS for secure transport• Handles Retransmissions of messages with specific interval• Scripted call generation and call reception• Supports 64-bit version to enhance signalling performance• Supports joining conference call, unattended call transfer, attended call transfer, call hold, auto call rejection, and silence packets generation• Ability to send "reliable provisional responses" and start early media actions• Ability to implement IP Spoofing for any network like Class C, Class B etc• Supports in dialog and out of dialog transactions for SUBSCRIBE, NOTIFY, OPTIONS, REFER and INFO SIP methods
Automation	<ul style="list-style-type: none">• Automation, Remote access, and Schedulers to run tests 24/7• Client-server model allows users to control all features of MAPS™ through APIs• Supported clients include Python and Java

MAPS™ SIP Highlights (Contd.)

Traffic	<ul style="list-style-type: none">• Supports various RTP traffic (PKS102) such as, digits, voice file, tones, IVR, FAX, and Video in IP networks• Supports almost all industry standard voice codec types - G.722, G.729, G.726, GSM, AMR, EVRC, EVS, OPUS, SMV, iLBC, SPEEX, and more. *AMR and EVRC variants require additional licenses• Supports 64-bit RTP core to enhance performance - handles increased call rate of up to 3000 calls with high volume traffic.• Supports both G.711 Pass Through Fax Simulation (PKS200) and T.38 Fax Simulation over UDPTL (PKS211)• Generates RTP-based video traffic using pre-encoded video payloads• Study packet effects through impairment generation:<ul style="list-style-type: none">• Latency (Uniform distributed and Normal distributed)• Packet loss (Periodic and Random)• Packet effects (Duplicate and Out of order)• Supports Secure Real-time Transport Protocol (or SRTP) traffic initialized over TLS (Transport Layer Security) or SSL (OpenSSL)• User-defined voice quality statistics for received RTP Traffic can be calculated and updated periodically during run-time to a csv file• Supports simulation of SIP/MSRP User Agents end-points in an NG9-1-1 network and send and receive communications over IP networks. MSRP sessions supports simulation of IM Only Calls, Audio and IM Calls, and Video and IM Call types
---------	---

SIP Call Types

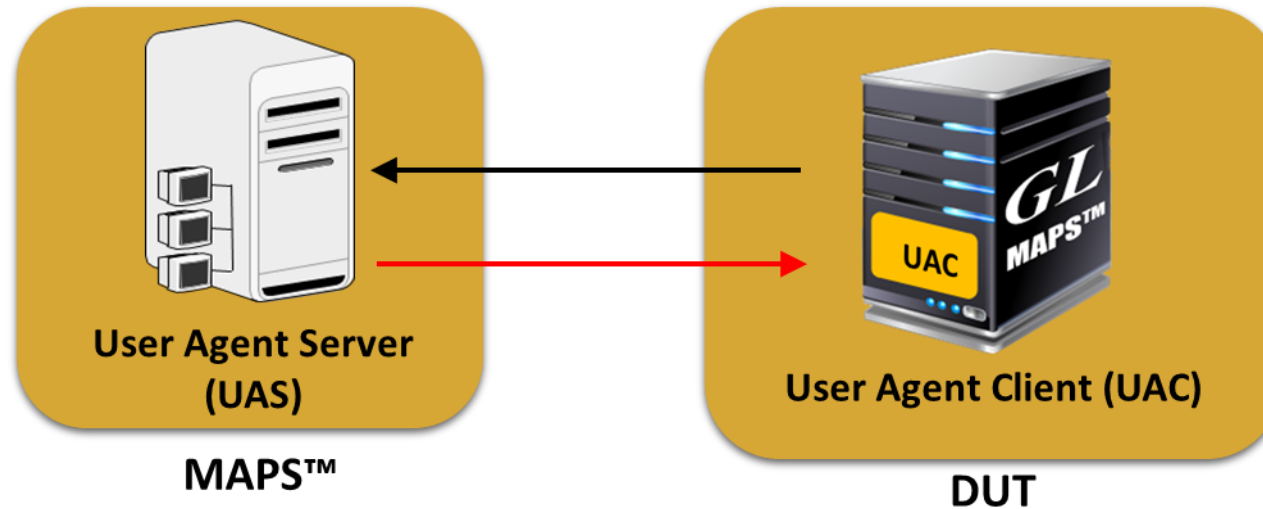
- Registration and Normal Call
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX) and Global Error (6xx)

MAPS™ SIP Configured as UAS

Testing UAC

Scenario: MAPS™ acting as UAS and testing UAC

- MAPS™ acting as UAS receives messages from UAC (DUT)
- DUT (UAC) generates SIP messages

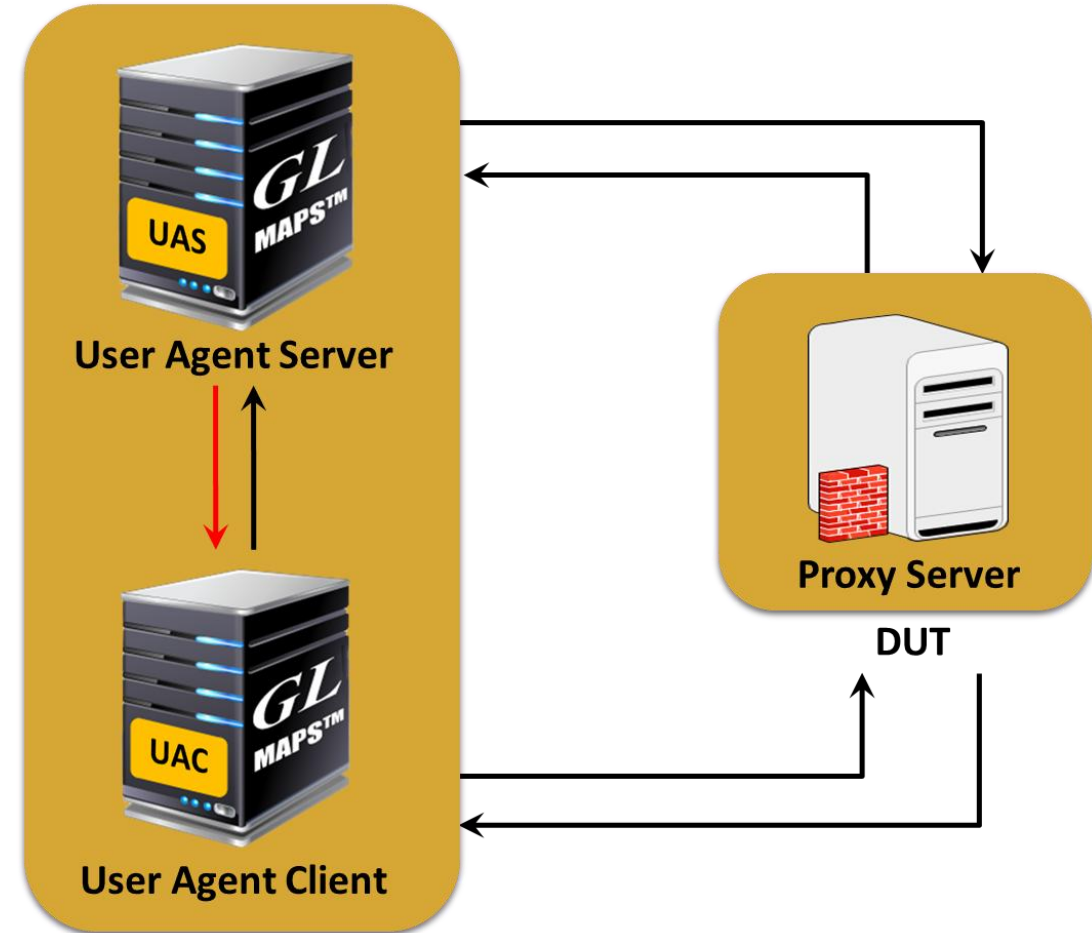


MAPS™ SIP Configured as UAC / UAS

Testing Proxy Server / B2B UA

Scenario: MAPS™ acting as UAS and UAC and testing Proxy.

- MAPS™ can be configured to act as UAC and UAS simultaneously so that entire Proxy testing can be automated

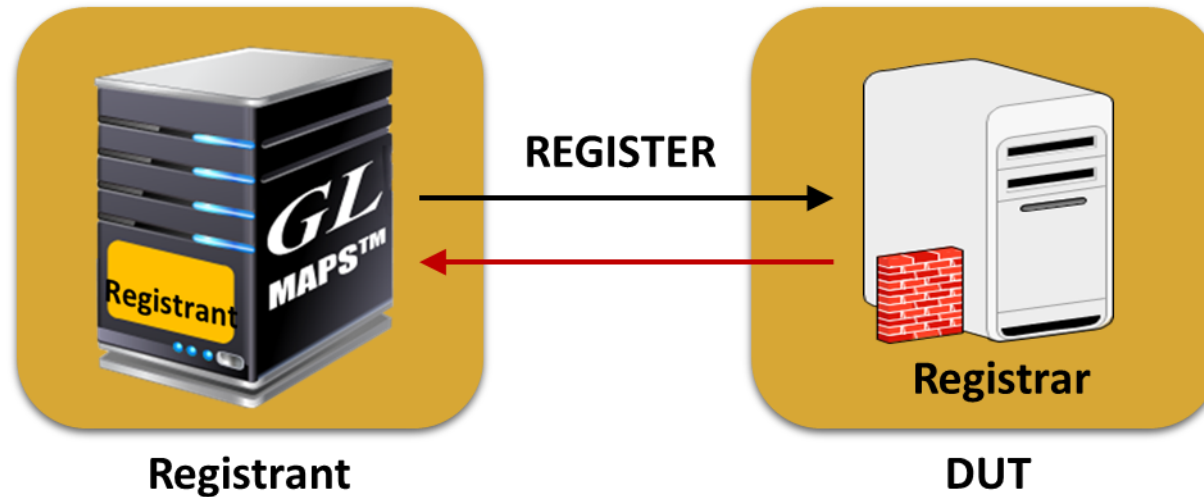


MAPS™ SIP Configured as Registrant

Testing Registrar

Scenario: MAPS™ acting as Registrant and testing Registrar.

- MAPS™ can be configured to act as Registrant and to generate registration request messages to automate the entire Registrar (DUT) testing

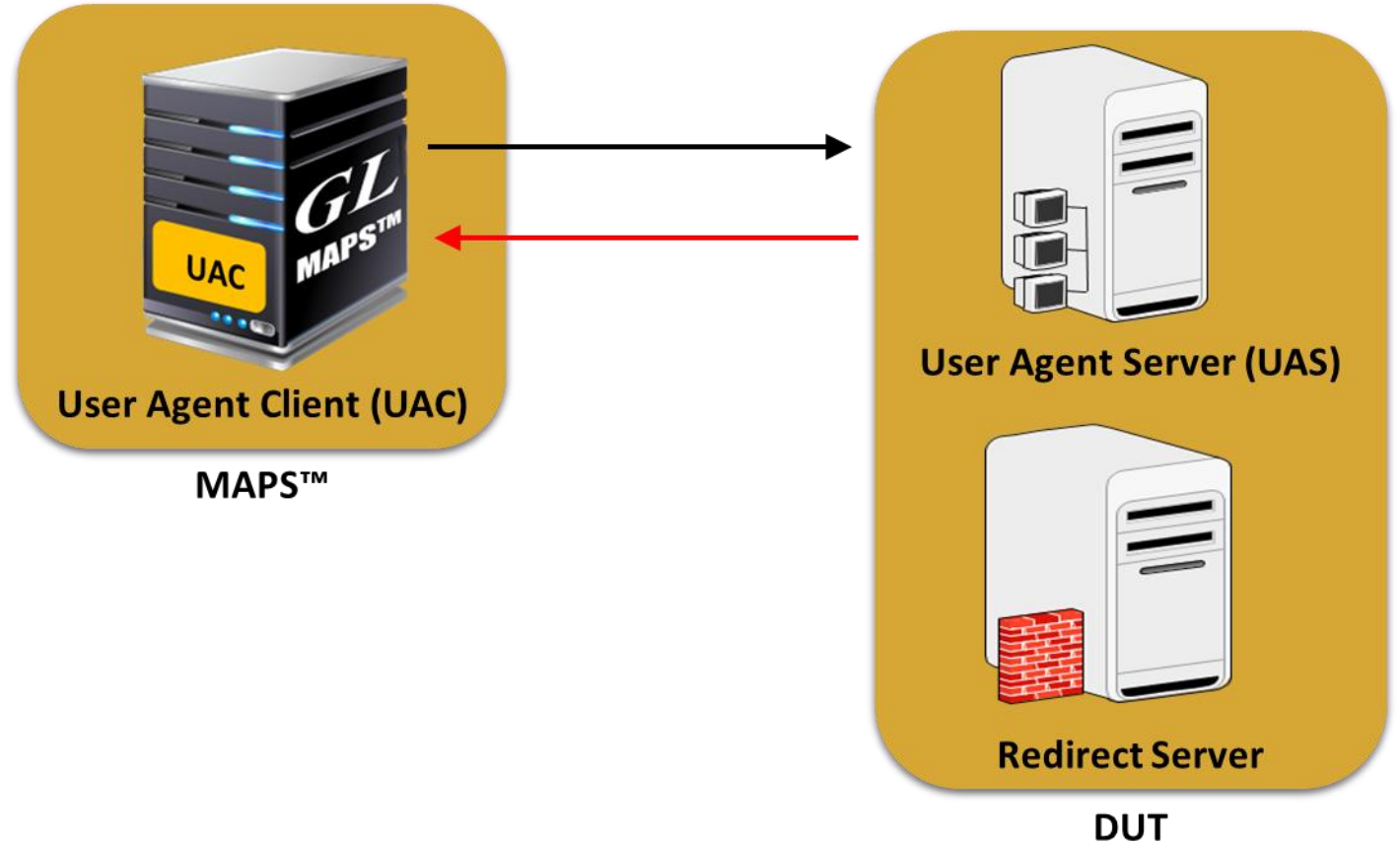


MAPS™ SIP Configured as UAC

Testing UAS & Redirect Server

Scenario: MAPS™ testing Redirect Server and / or UAS

- MAPS™ can be configured to act as UAC & generate SIP messages
- Tests Redirect Server and /or UAS; allows redirection of call scenarios to be automated

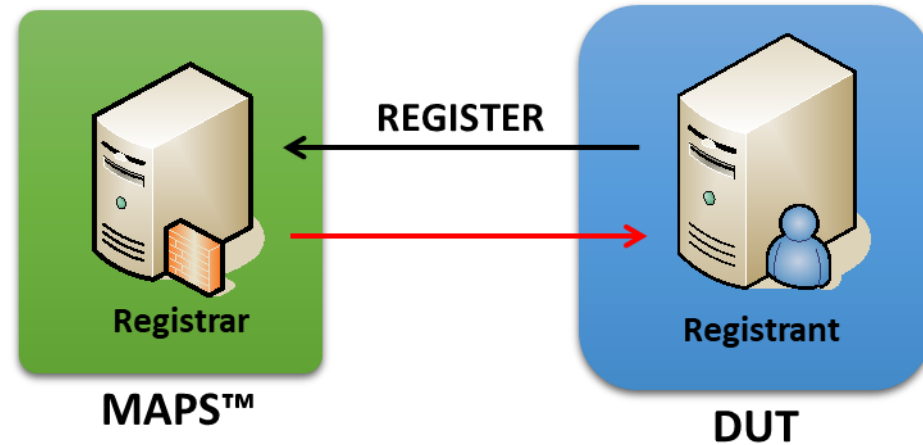


MAPS™ SIP Configured as Registrar

Testing Registrant

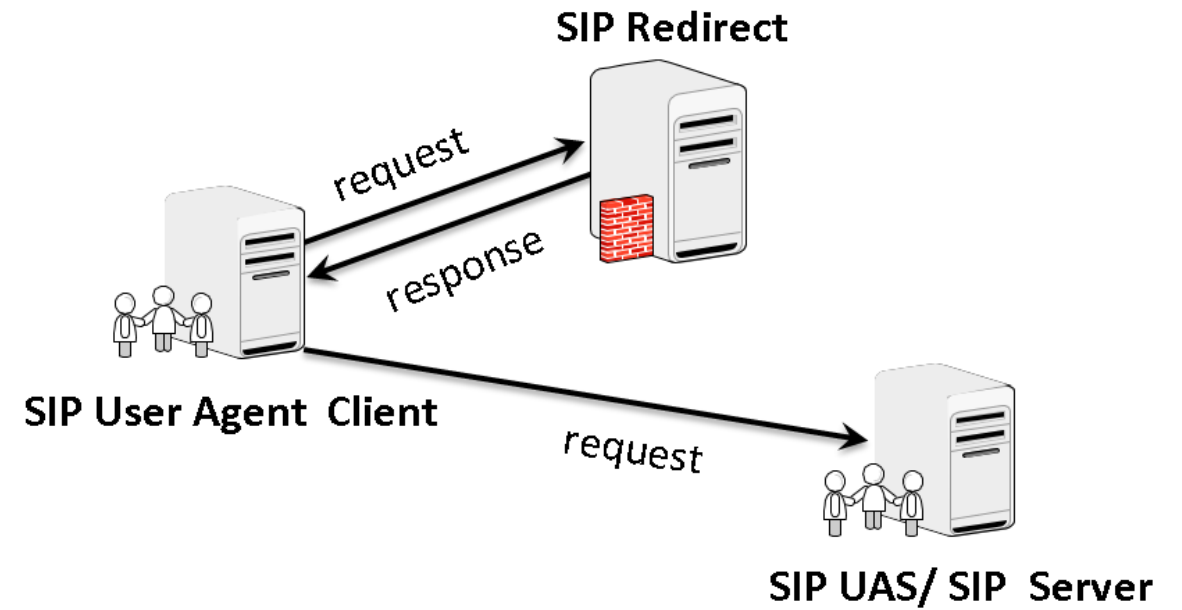
Scenario: MAPS™ acting as Registrar and testing Registrant

- MAPS™ acts as Registrar and processes received registration request messages from Registrant (DUT) while conforming Registrant
- DUT (Registrant) generates REGISTRATION SIP messages



SIP Redirect Server

- Returns the next address to originator instead of forwarding
- Originator retries with the new address



Call Generation (UAC)

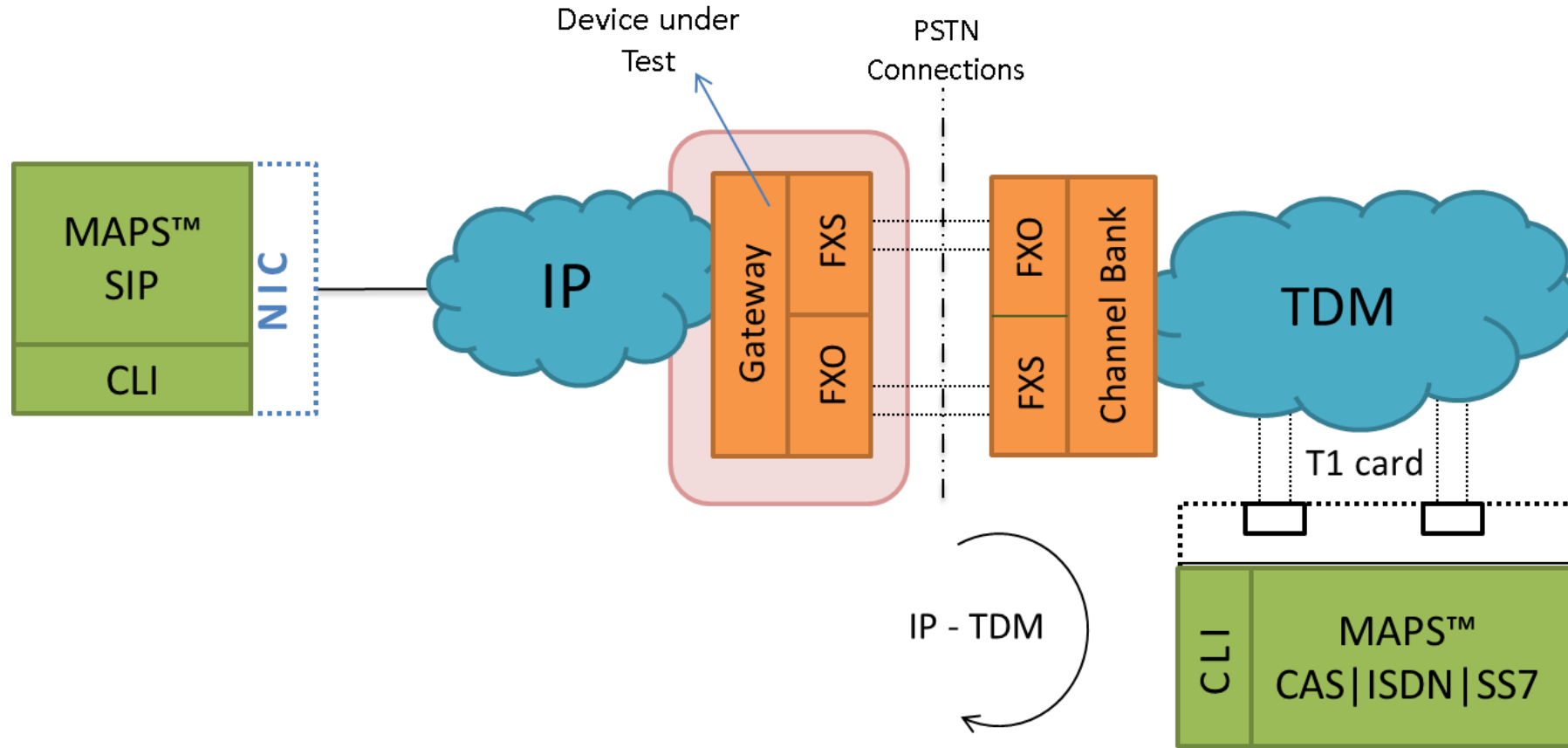
- Registrant – Registers with Registrar
- Call with Auto Traffic of RTP Action
- Traffic Impairments
- Simulates IVR (Interactive Voice Response) for RTP traffic
- Call through Proxy
- Sequential and Random Generation of Calls
- Simultaneous Generation of Calls
- Load Generation (Stress Testing)

Call Reception (UAS)

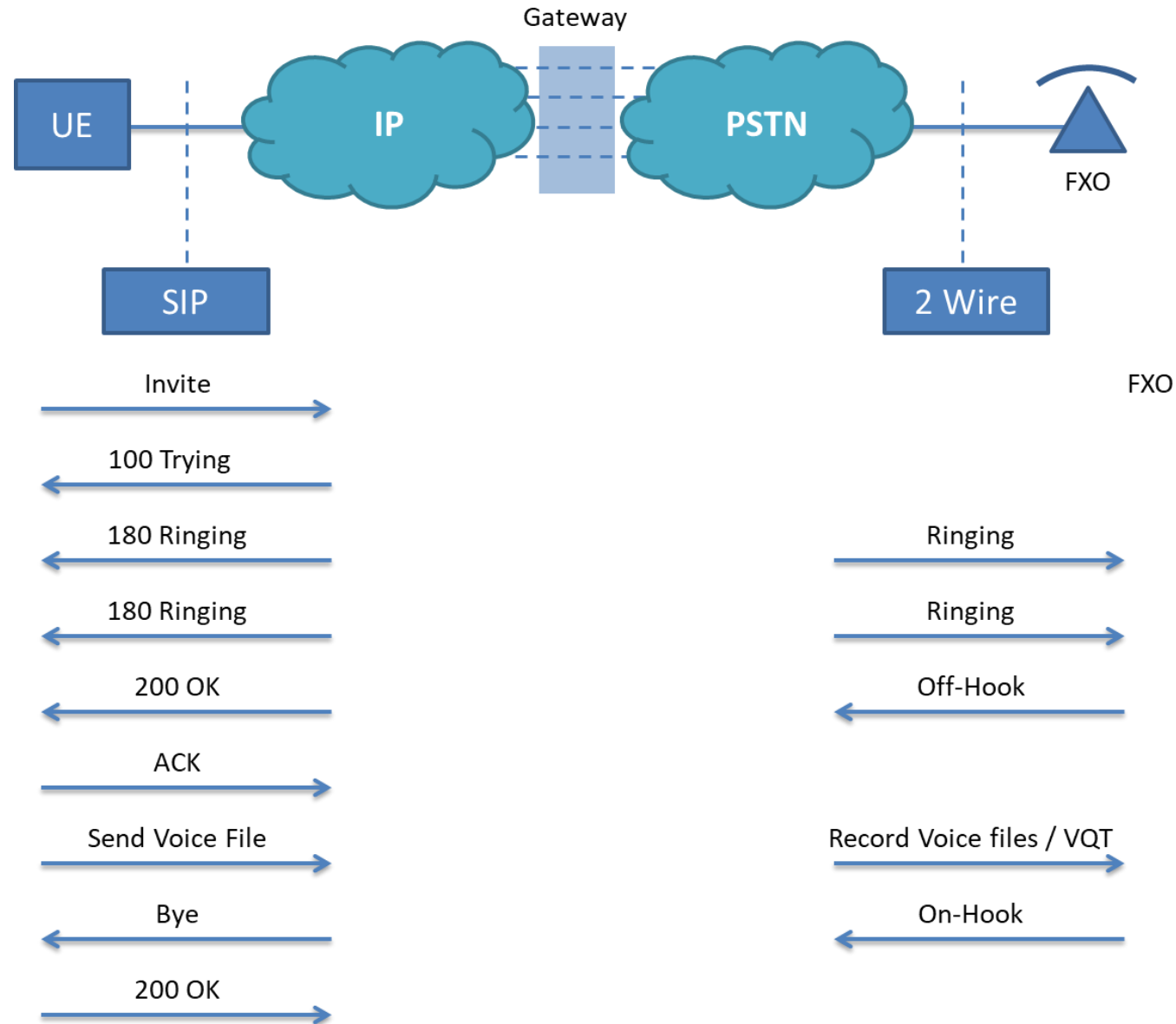
- Registrar – Accepts the registration from registrant
- Call Redirection – Redirect the call to new location
- Call Transfer - Transfers the call using REFER Method
- Authentication – Challenging the incoming message for credential
- Early Media (PRACK support)
- Rejecting the call with Client Error (4XX), Server Error (5XX), and Global Error (6xx)

End-to-End Gateway Testing

- Evaluates Gateway / ATA product features such as call connectivity, call signaling, traffic generation, voice quality testing, codec, and hundreds of other features

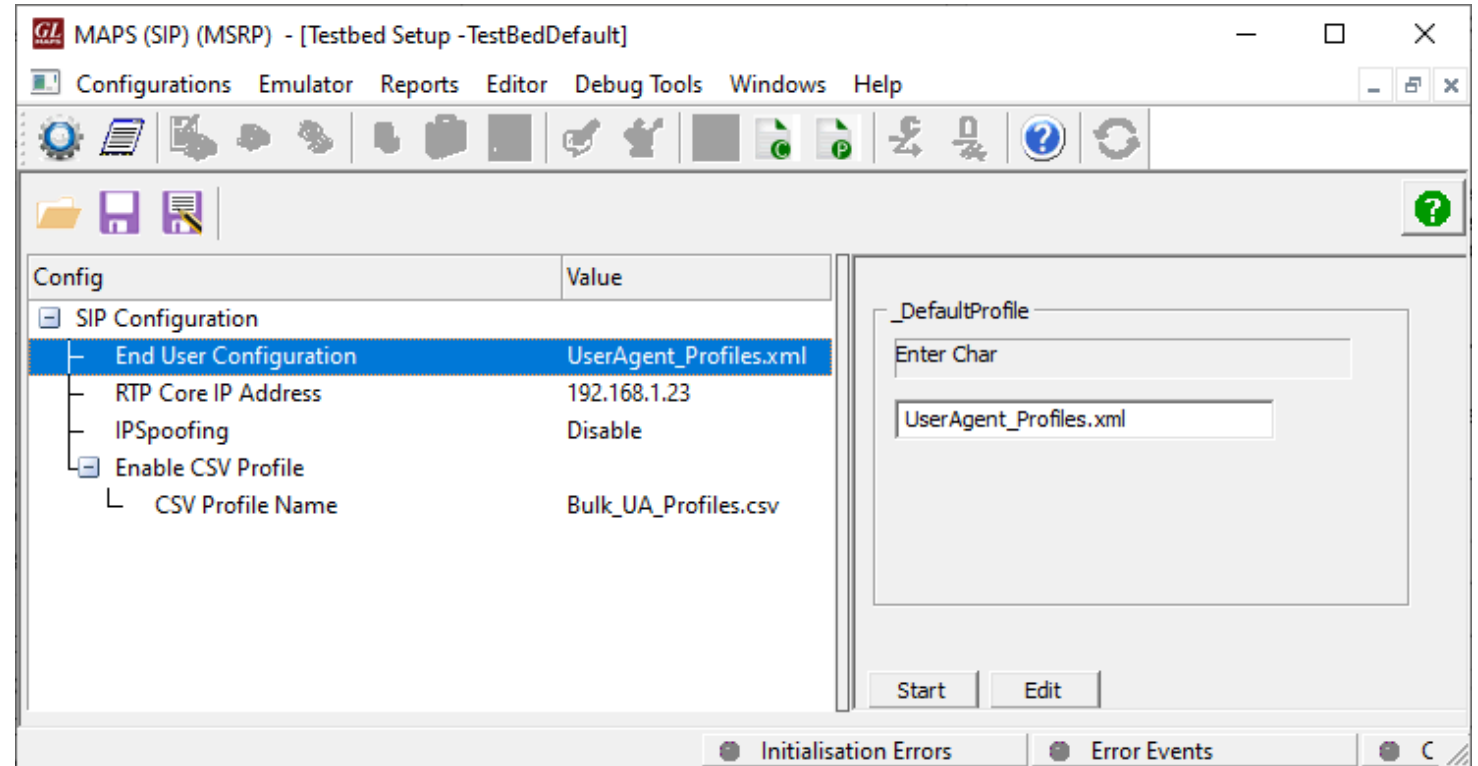


End-to-End Gateway Testing Call Scenario



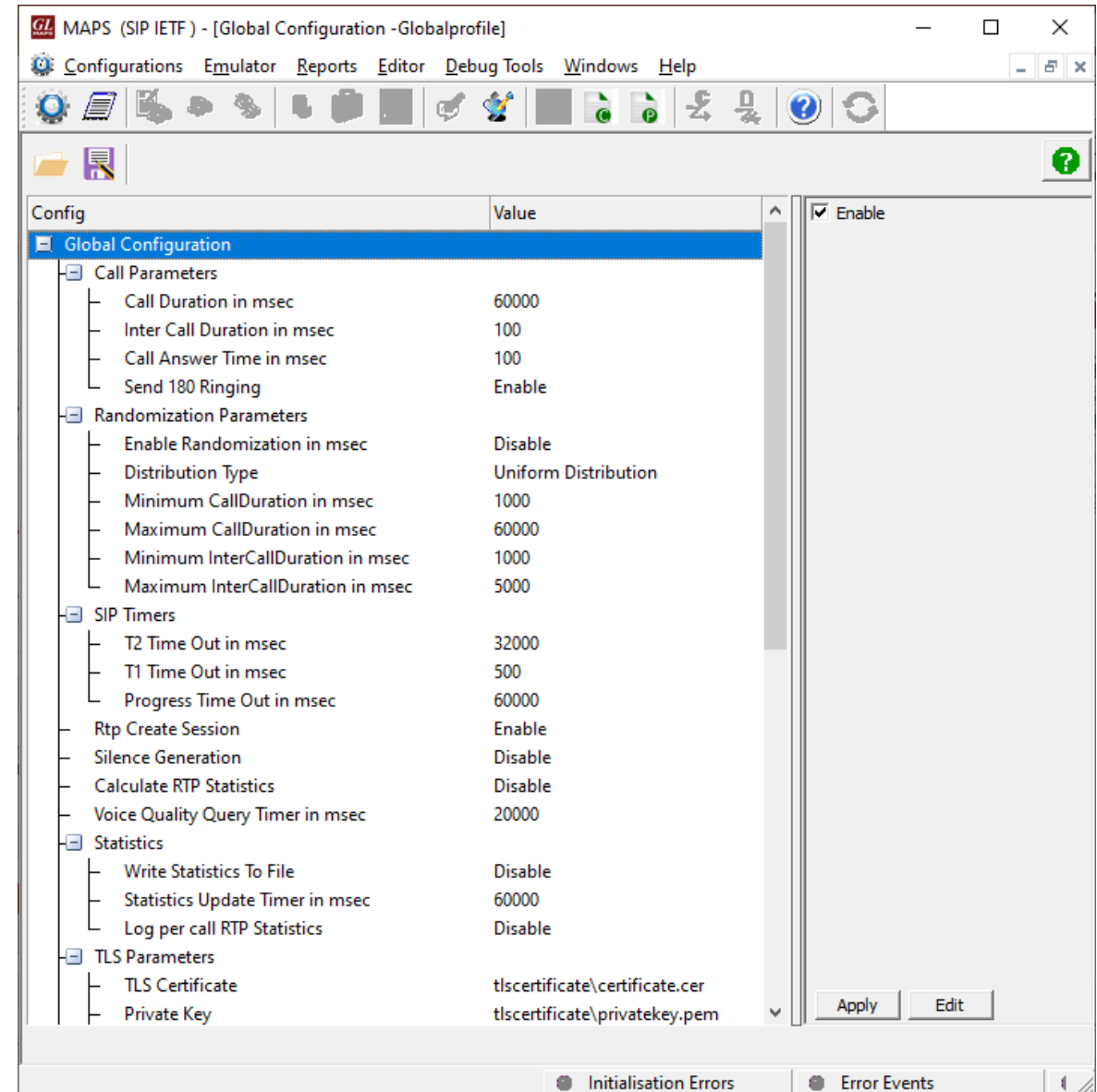
Test Bed Configuration

- **End User Configuration:** xml file containing one or more endpoint configurations
- **RTP Core IP Address:** IP Address of the system on which the RTP Core should be invoked
- **IP Spoofing:** permits user to assign one or more virtual IP addresses to NIC



Global Configuration

- A list of variables/values that are automatically declared and assigned at the start of any script execution
- A script may locally override the values assigned here
- A script may also ignore these variables entirely. For example, Call Duration is not a hard limit on the length of a call, it is just a variable the script may use



The screenshot shows the 'Global Configuration' window in the MAPS (SIP IETF) application. The window title is 'MAPS (SIP IETF) - [Global Configuration - Globalprofile]'. The menu bar includes 'Configurations', 'Emulator', 'Reports', 'Editor', 'Debug Tools', 'Windows', and 'Help'. The toolbar contains various icons for file operations and settings. The main area is a table with columns 'Config', 'Value', and 'Enable'. The 'Global Configuration' section is expanded, showing several sub-sections: 'Call Parameters', 'Randomization Parameters', 'SIP Timers', 'Statistics', and 'TLS Parameters'. Each sub-section contains a list of parameters with their corresponding values and an 'Enable' checkbox.

Config	Value	Enable
Global Configuration		<input checked="" type="checkbox"/>
Call Parameters		
Call Duration in msec	60000	
Inter Call Duration in msec	100	
Call Answer Time in msec	100	
Send 180 Ringing	Enable	
Randomization Parameters		
Enable Randomization in msec	Disable	
Distribution Type	Uniform Distribution	
Minimum CallDuration in msec	1000	
Maximum CallDuration in msec	60000	
Minimum InterCallDuration in msec	1000	
Maximum InterCallDuration in msec	5000	
SIP Timers		
T2 Time Out in msec	32000	
T1 Time Out in msec	500	
Progress Time Out in msec	60000	
Rtp Create Session	Enable	
Silence Generation	Disable	
Calculate RTP Statistics	Disable	
Voice Quality Query Timer in msec	20000	
Statistics		
Write Statistics To File	Disable	
Statistics Update Timer in msec	60000	
Log per call RTP Statistics	Disable	
TLS Parameters		
TLS Certificate	tlscertificate\certificate.cer	
Private Key	tlscertificate\privatekey.pem	

Buttons: Apply, Edit

Bottom status bar: Initialisation Errors, Error Events

User Agents Configuration

- Each Profile Group contains one or several sub-profiles
- Each sub-profile is a set of variables which together define a single SIP Endpoint
- Not every field in a profile is relevant to every script execution
- Profile Editor has a “Quick Config” tool to help users create multiple different sub-profiles

MAPS (SIP IETF) - [Profile Editor - UserAgent_Profiles]

Configurations Emulator Reports Editor Debug Tools Windows Help

Profiles (Edit-F2)

#	Profiles (Edit-F2)
1	Profile0001
2	Profile0002
3	Profile0003
4	Profile0004
5	Profile0005
6	Profile0006
7	Profile0007
8	Profile0008
9	Profile0009
10	Profile0010

Config Value

Profile0001

- Apply DiffServ Code Point
- Proxy Parameters
- OPTIONS Parameters
- Call Parameters
 - IP Address Type: IPv4
 - Transport: UDP
 - Call Type: AudioCall
 - Contact Address: 0001@192.168.12.74
 - Address Of Record: 0001@192.168.12.74
 - To Address: 0001@192.168.12.78
 - Subnet Mask: 255.255.255.0
 - Cipher Suite for TLS: ALL
 - SRTP: Disable
 - SRTP Algorithm: AES_CM_128_HMAC_SH...
 - Local Call Duration in msec: 0
 - Local Call Answer Time in msec: 0
- MSRP Parameters
 - MSRP IP Address: 192.168.12.78
 - MSRP Type: TCP/MSRP
 - MSRP AcceptType List: text/plain
 - MSRP HeartBeat Timeout in msec: 0
- SDP Parameters
 - RTP IP Address: 192.168.12.74
 - Packetization time in msec: 20
- SMS Call Parameters
 - SMS Character Set: Default
 - SMS Data for Default and 8 Bit Data: GL@005

Enable

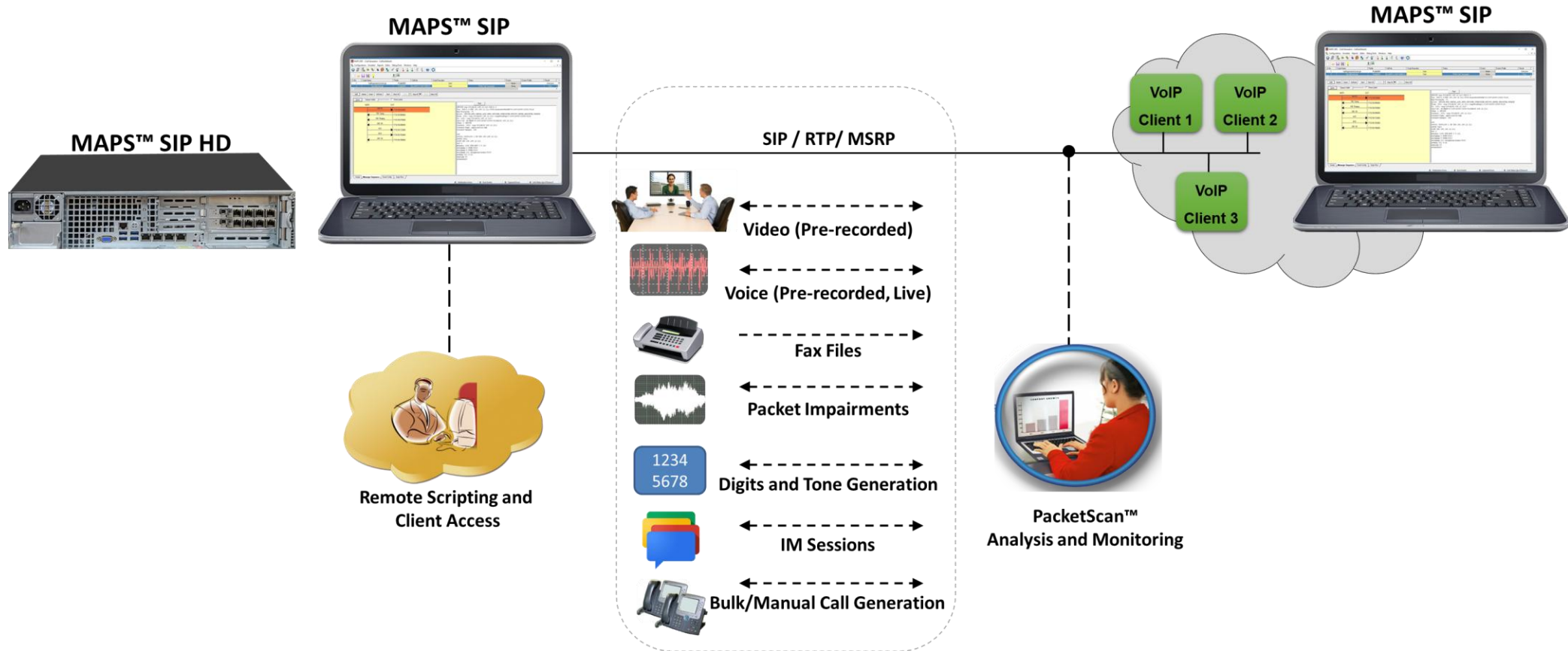
Add Insert Delete Properties

Insert Delete Clear

Initialisation Errors Error Events Captured Errors

IP Traffic Simulation Capabilities and Performance

2000+ Simultaneous Calls (SIP + RTP Voice)
500 Simultaneous Calls (SIP + RTP Video)
500 Simultaneous Calls (SIP + IM MSRP)



SIP Capabilities and Performance

Product Version	Max Simultaneous Calls			
	Only Signaling	Signaling + RTP Voice Traffic	Signaling + RTP Video Traffic	Signaling + MSRP (IM) Traffic
MAPS™ SIP 64-bit (Core i7 with 12GB RAM)	30,000 Calls @ 250 CPS	2500 @ 250 CPS	500	500
MAPS™ SIP HD 64-bit (Zeon Server with 16 Processors and 64GB RAM)	100,000 Calls @250 CPS	20000 @ 250 CPS	-	-

Call Generation with Voice Traffic

GL MAPS (SIP) - [Call Generation -CallGenDefault]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown
2	SipCallControl.gls	Profile0001	GL-MAPS-3-104714086-10...	Start	PCMU Call Terminated	None		Pass

Column Width Show Latest

MAPS	DUT
INVITE	17:52:58.529000
100 Trying	17:52:58.558000
180 Ringing	17:52:58.570000
200 OK	17:52:58.696000
ACK	17:52:58.723000
BYE	17:53:58.762000
200 OK	17:53:58.788000

Find

```

INVITE sip:0001@192.168.12.210 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.211:5060;branch=z9hG4bK-4-104714086-10291-9124
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER, UPDATE
From: 0001 <sip:0001@192.168.12.211>;tag=FromTag-1-104714086-10288-9124
To: 0001 <sip:0001@192.168.12.210>
Call-ID: GL-MAPS-3-104714086-10290-9124@192.168.12.211
CSeq: 1 INVITE
Contact: 0001 <sip:0001@192.168.12.211>
Content-Type: application/sdp
Content-Length: 240

v=0
o=0001 32891133 1 IN IP4 192.168.12.211
s=SIP Call
c=IN IP4 192.168.12.211
t=0 0
m=audio 1030 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
  
```

Initialisation Errors
 Error Events
 Captured Errors
 Link Status Up=0 Down=0

Call Generation with IVR Traffic

The screenshot displays the MAPS (SIP) - [Call Reception] application window. At the top, there is a menu bar with options: Configurations, Emulator, Reports, Editor, Debug Tools, Windows, and Help. Below the menu is a toolbar with various icons. The main area is divided into a table at the top and a detailed view below.

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Results
1	SipCallControl.gls	Profile0001	GL-MAPS-3-104714086-10290-912..	Completed	PCMU Call Terminated	None		Pass

Below the table, there are control buttons: Stop, Stop All, Abort, Abort All, Show Records (checked), Select Active Call, Auto Trash, and Trash. Below these are Save, Column Width, and Show Latest options.

The central part of the interface shows a message sequence diagram between DUT and MAPS. The diagram is as follows:

```
graph LR
    DUT -- INVITE --> MAPS[17:52:58.539000]
    MAPS -- 100 Trying --> DUT[17:52:58.549000]
    MAPS -- 180 Ringing --> DUT[17:52:58.560000]
    MAPS -- 200 OK --> DUT[17:52:58.681000]
    DUT -- ACK --> MAPS[17:52:58.736000]
    DUT -- BYE --> MAPS[17:53:58.768000]
    MAPS -- 200 OK --> DUT[17:53:58.776000]
```

To the right of the diagram is a text area containing SIP message details:

```
INVITE sip:0001@192.168.12.210 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.211:5060;branch=z9hG4bK-4-104714086-10291-9124
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER, UPDATE
From: 0001 <sip:0001@192.168.12.211>;tag=FromTag-1-104714086-10288-9124
To: 0001 <sip:0001@192.168.12.210>
Call-ID: GL-MAPS-3-104714086-10290-9124@192.168.12.211
CSeq: 1 INVITE
Contact: 0001 <sip:0001@192.168.12.211>
Content-Type: application/sdp
Content-Length: 240

v=0
o=0001 32891133 1 IN IP4 192.168.12.211
s=SIP Call
c=IN IP4 192.168.12.211
t=0 0
m=audio 1030 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

At the bottom of the window, there are tabs for Scripts, Message Sequence (selected), Event Config, and Script Flow. The status bar at the very bottom shows: Initialisation Errors, Error Events, Captured Errors, and Link Status Up=0 Down=0.

RTP Voice Quality Measurements

- RTP based Voice Quality (MOS and R-Factor) measurement are calculated and updated periodically for the received streams
- Call quality metrics includes Listening MOS, Conversational MOS, Packet Loss, Discarded Packets, Out of Sequence Packets, Duplicate Packets, Delay and Jitter

Name	Values
Active RTP Sessions	0
Completed RTP Sessions	6
Sessions With Zero Receive Traffic	0
-----	0
MOS Score Stats	0
-----	0
Sessions with Mos (5.0 - 4.0)	4 [66%]
Sessions with Mos (4.0 - 3.0)	0 [0%]
Sessions with Mos (3.0 - 2.0)	0 [0%]
Sessions with Mos (< 2.0)	0 [0%]
-----	0
Total RTP Packet Sent	1597
Total RTP Packet Received	2097
-----	0
Packet-Loss Stats	0
-----	0
Total PacketLoss	0 [0%]
Sessions with Zero Packet-Loss	4 [66%]
Sessions with Packet-Loss(<1%)	0 [0%]
Sessions with Packet-Loss(1% - 5%)	0 [0%]
Sessions with Packet-Loss(5% - 10%)	0 [0%]
Sessions with Packet-Loss(>10%)	0 [0%]
-----	0
Packet-Discarded Stats	0
-----	0
Total PacketDiscarded	0 [0%]
Sessions with Zero Packet-Discard	4 [66%]
Sessions with Packet-Discard(<1%)	0 [0%]
Sessions with Packet-Discard(1% - 5%)	0 [0%]
Sessions with Packet-Discard(5% - 10%)	0 [0%]
Sessions with Packet-Discard(>10%)	0 [0%]
-----	0
Packet-Duplicate Stats	0
-----	0
Total Duplicate Packet	0 [0%]
Sessions with Zero Duplicate Packets	4 [66%]
Sessions with Duplicate Packets(<1%)	0 [0%]

Event Log, Error Events, Captured Errors

The image displays three overlapping screenshots of the 'Events' application window, illustrating different views of system events and errors.

Top Screenshot: Event Log View

The 'Event Log' tab is selected. The table shows the following data:

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2024-9-23 13:06:42.359000	Script Initialized	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.393000	TransportId = 1	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.393000	TransportStatus =	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.393000	MediaPAddress=192.168.12.216, AudioMediaPort=1038	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.395000	RtpCoreSrtpAlgorithm =	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.395000	RtpCoreSrtpKey =	CGProtScriptId-10-1006011713-9549-14080	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.405000	INVITE Sent		SIP-Protocol.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.420000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.420000	PROGRESS Received	GL-MAPS-18-1006011716-9553-7856@192.168.12.216	SipCallControl.gls	CGProtScriptId-10-1006011713-9549-14080
2024-9-23 13:06:42.427000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId-10-1006011713-9549-14080

Middle Screenshot: Error Events View

The 'Error Events' tab is selected. The table shows the following data:

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2024-9-23 12:58:37.078000	Session Creation Failed	CGProtScriptId-3-1005519694-9487-14080	SipCallControl.gls	CGProtScriptId-3-1005519694-9487-14080
2024-9-23 13:00:01.136000	Session Creation Failed	CGProtScriptId-4-1005603631-9496-14080	SipCallControl.gls	CGProtScriptId-4-1005603631-9496-14080
2024-9-23 13:00:01.237000	4xx Failure Message=480 INVITE	GL-MAPS-10-1005605428-9501-14132@192.168.12.216	SipCallControl.gls	CGProtScriptId-4-1005603631-9496-14080
2024-9-23 13:01:05.296000	Session Creation Failed	CGProtScriptId-5-1005669593-9505-14080	SipCallControl.gls	CGProtScriptId-5-1005669593-9505-14080
2024-9-23 13:03:37.277000	6xx Failure Message=606 INVITE	GL-MAPS-10-1005826538-9530-13992@192.168.12.216	SipCallControl.gls	CGProtScriptId-7-1005826534-9526-14080
2024-9-23 13:05:06.385000	Retransmission Time Out	GL-MAPS-9-1005884067-9537-23124@192.168.12.216	SipCallControl.gls	CGProtScriptId-8-1005882418-9532-14080
2024-9-23 13:05:06.385000	Retransmission Timeout for the Message = INVITE	GL-MAPS-9-1005884067-9537-23124@192.168.12.216	SipCallControl.gls	CGProtScriptId-8-1005882418-9532-14080

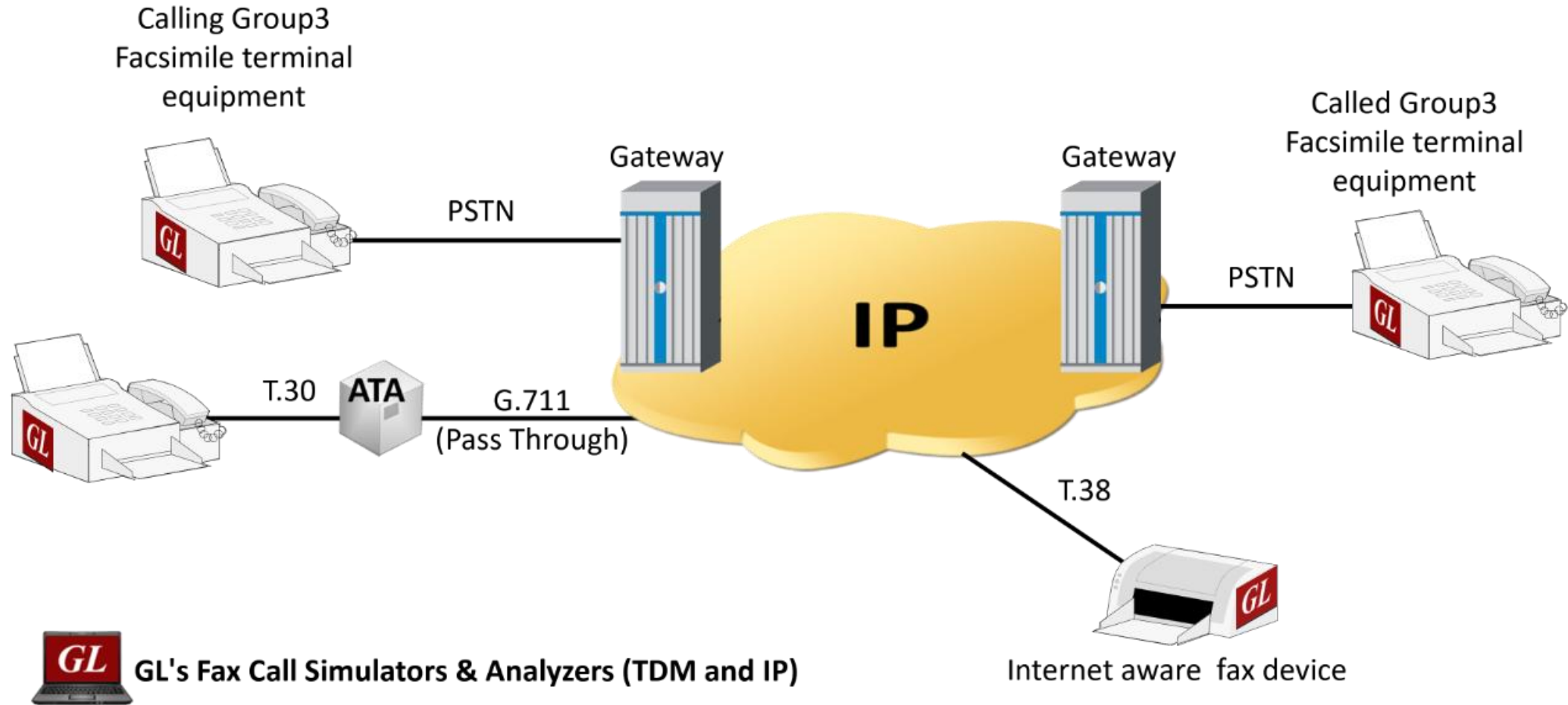
Bottom Screenshot: Captured Errors View

The 'Captured Errors' tab is selected. The table shows the following data:

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2024-9-23 12:57:27.877000	Error::Creating UDP Socket Failed due to Errorcode=10049			
2024-9-23 12:57:27.878000	192.168.12.200 : 5060 : please verify IP/Port not in use			
2024-9-23 12:57:27.879000	192.168.12.200 : 5061 : please verify IP/Port not in use			
2024-9-23 12:58:32.060000	RtpCore Failure Response : Create Session Failed:RTP_CAUSE_INERNAL_ERR	CGProtScriptId-3-1005519694-9487-14080	SipCallControl.gls	CGProtScriptId-3-1005519694-9487-14080
2024-9-23 12:59:56.116000	RtpCore Failure Response : Create Session Failed:RTP_CAUSE_INERNAL_ERR	CGProtScriptId-4-1005603631-9496-14080	SipCallControl.gls	CGProtScriptId-4-1005603631-9496-14080
2024-9-23 13:01:00.291000	RtpCore Failure Response : Create Session Failed:RTP_CAUSE_INERNAL_ERR	CGProtScriptId-5-1005669593-9505-14080	SipCallControl.gls	CGProtScriptId-5-1005669593-9505-14080

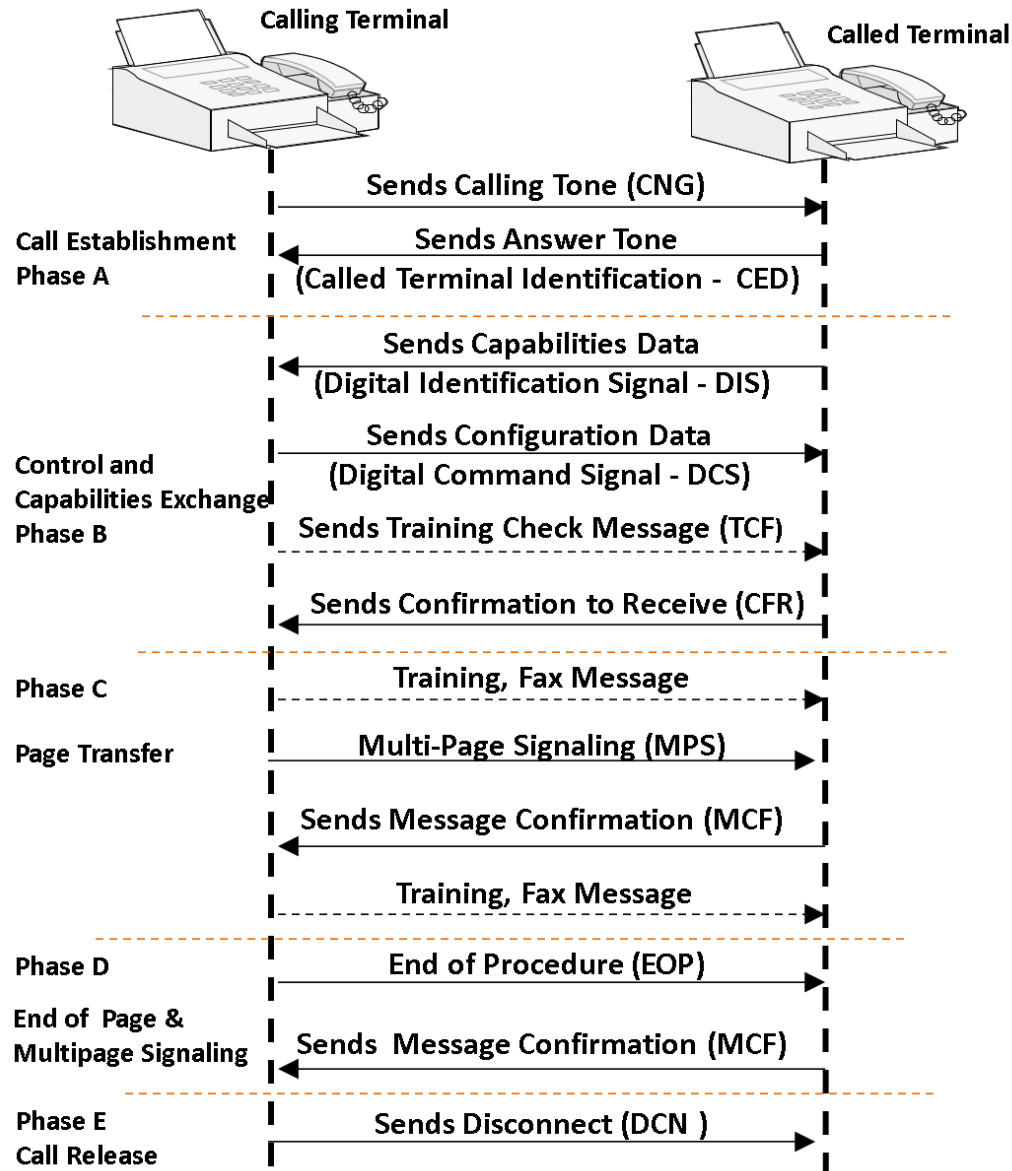
Below the table in the bottom screenshot, there is a 'Save Events' section with a 'Clear' button and a checkbox labeled 'Capture Events to file' followed by a file selection button.

Fax Simulation over IP

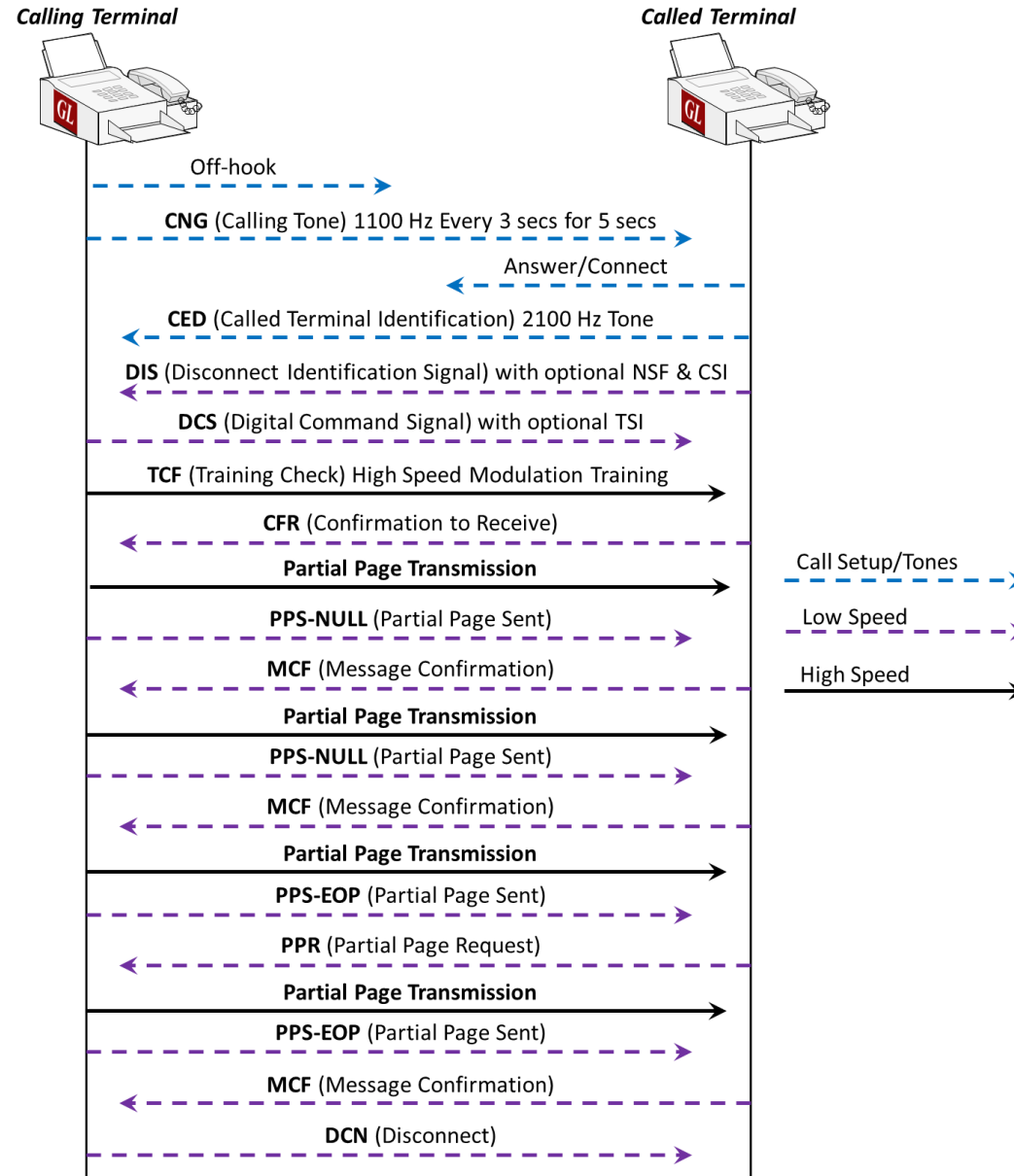


- RTP G.711 Pass Through Fax Simulation (PKS200)
- T.38 Fax Simulation over UDPTL (PKS211)

Call Scenarios - Fax T.30



T.38 Fax Emulation over IP using MAPS™



T.38 Fax Call in Progress and Related Events

The screenshot displays the GL MAPS (Message Automation Protocol Simulation) interface, showing a T.38 Fax Call in Progress and related events.

Call Information:

- Script Name: SipCallControl.gls
- Profile: Profile001
- Call Info: GL-MAPS_3_775735732-4923-3768@192.168.12.212
- Script Execution: Stop
- Status: Fax Session Successful
- Events: SIP_TerminateCall

Message Sequence (MAPS vs DUT):

- 11:12:41.097000: INVITE
- 11:12:41.123000: 100 Trying
- 11:12:41.129000: 180 Ringing
- 11:12:41.248000: 200 OK
- 11:12:41.260000: ACK
- 11:12:41.299000: INVITE (highlighted in orange)
- 11:12:41.306000: 200 OK
- 11:12:41.308000: ACK
- 11:13:15.904000: 33600 Rate of V34 selected after
- 11:13:15.905000: CSI(Called Subscriber Identification)
- 11:13:15.906000: DIS(Digital Identification Signal)
- 11:13:15.906000: EGM mode Selected in DCS
- 11:13:15.907000: MMR Encoding selected in DCS

Event Log (Captured Events):

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2017-10-26 11:12:41.132000	PROGRESS Received	GL-MAPS_3_775735732-49...	SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.132000	PROGRESS Received	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.260000	ACK Sent	GL-MAPS_3_775735732-49...	SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.261000	Call Connected	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:12:41.308000	ACK Sent	GL-MAPS_3_775735732-49...	SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.904000	Fax - Status: 33600 Rate of V34 selected after MPH exchange	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.905000	Fax - Status: CSI(Called Subscriber Identification)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.906000	Fax - Status: DIS(Digital Identification Signal)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.906000	Fax - Status: ECM mode Selected in DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.907000	Fax - Status: MMR Encoding selected in DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.908000	Fax - Status: 200x200 Resolution selected in the DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.908000	Fax - Status: A4 pagesize selected in the DCS	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.909000	Fax - Status: TSI(Transmitting Subscriber Identification)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.909000	Fax - Status: DCS(Digital Command Signal)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.910000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.911000	Fax - Status: Transmitter Started To Train	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.911000	Fax - Status: Transmitter Train Successful	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.912000	Fax - Status: CFR(Confirmation To Receive)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.912000	Fax - Status: Image Transmit Start	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.913000	Fax - Status: Image Transmit End	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.913000	Fax - Status: PPS NULL(Current Partial Page Block: Transmission Complete)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.914000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.914000	Fax - Status: MCFR(Message Confirmation)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.915000	Fax - Status: Image Transmit Start	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.915000	Fax - Status: Image Transmit End	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.916000	Fax - Status: PPS EOP(All Pages Transmitted)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.916000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.917000	Fax - Status: MCFR(Message Confirmation)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.917000	Fax - Status: DCN(Disconnect)	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.918000	Fax - Status: V21 Signal Done	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.919000	Fax Session Successful	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:13:15.919000	Fax - Status: FaxSessionDuration = 2898 msecInitialModem = "V34" InitialRate = "33600" FinalMod...	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.280000	BYE Sent	GL-MAPS_3_775735732-49...	SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.290000	200 Ok to BYE Received	GL-MAPS_3_775735732-49...	SIP-Protocol.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.290000	Call Terminated	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...
2017-10-26 11:14:41.290000	Inter Call Duration = 100	GL-MAPS_3_775735732-49...	SipCallControl.gls	CGProtScriptId_1_77573572...

Call Content:

```

INVITE sip:0001@192.168.12.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.212:5060;bran
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTION
From: 0001 <sip:0001@192.168.12.212>;tag
To: 0001 <sip:0001@192.168.12.213>;tag=T
Call-ID: GL-MAPS_3_775735732-4923-3768@1
CSeq: 2 INVITE
Contact: 0001 <sip:0001@192.168.12.212>
Supported: 100rel
Content-Type: application/sdp
Content-Length: 361

v=0
o=0001 33852938 33852938 IN IP4 192.168.
s=SIP Call
c=IN IP4 192.168.12.212
t=0 0
m=image 1030 udpt1 t38
a=T38FaxVersion:3
a=T38FaxMaxBitRate:33600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCP
a=T38FaxMaxBuffer:400
a=T38FaxMaxDatagram:280
a=T38FaxUpDcEc:t38UDPRedundancy
    
```

Call Generation with FAX Traffic

The screenshot displays the GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation] interface, showing two windows illustrating call generation with FAX traffic.

Left Window: MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation]

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile
1	SipCallControl.gls	Profile0001	GL-MAPS_3_776162744-4937-3896@192.168.12.212	Stop	Fax Session Successful	SIP_TerminateCall	

Message Sequence (Left Window):

Direction	Message	Time
MAPS → DUT	INVITE	11:19:48.114000
DUT → MAPS	100 Trying	11:19:48.140000
DUT → MAPS	180 Ringing	11:19:48.145000
DUT → MAPS	200 OK	11:19:48.268000
DUT → MAPS	ACK	11:19:48.280000
DUT → MAPS	Fax Status :: Send Fax Started	11:19:48.343000
DUT → MAPS	33600 Rate of V34 selected after ...	11:20:22.163000
DUT → MAPS	V21 Signal Done	11:20:22.164000
DUT → MAPS	CSI(Called Subscriber Identification)	11:20:22.164000
DUT → MAPS	DIS(Digital Identification Signal)	11:20:22.165000
DUT → MAPS	ECM mode Selected in DCS	11:20:22.166000
DUT → MAPS	MMR Encoding selected in DCS	11:20:22.166000
DUT → MAPS	200x200 Resolution selected in th...	11:20:22.167000

Right Window: MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation - CallGenDefault]

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events	Result	Tot
1	SipRegistrationControl.gls	Profile0001		Start		None		Unknown	
2	SipCallControl.gls	Profile0003	GL-MAPS_1_14830463-305-3768@192.168.1.141	Stop	Fax Session Created	SIP_TerminateCall		Pass	

Message Sequence (Right Window):

Direction	Message	Time
MAPS → DUT	INVITE	16:58:18.881000
DUT → MAPS	100 Trying	16:58:19.244000
DUT → MAPS	180 Ringing	16:58:19.247000
DUT → MAPS	200 OK	16:58:19.361000
DUT → MAPS	ACK	16:58:19.369000
DUT → MAPS	INVITE	16:58:19.377000
DUT → MAPS	Fax Status :: 33600_Rate_of_V34_selected_after_MP...	16:58:19.379000
DUT → MAPS	33600_Rate_of_V34_selected_after_MPh_exchange	16:58:19.379000
DUT → MAPS	Fax Status :: CSI(Called_Subscriber_Identification)	16:58:19.379000
DUT → MAPS	CSI(Called_Subscriber_Identification)	16:58:19.379000
DUT → MAPS	Fax Status :: DIS(Digital_Identification_Signal)	16:58:19.379000

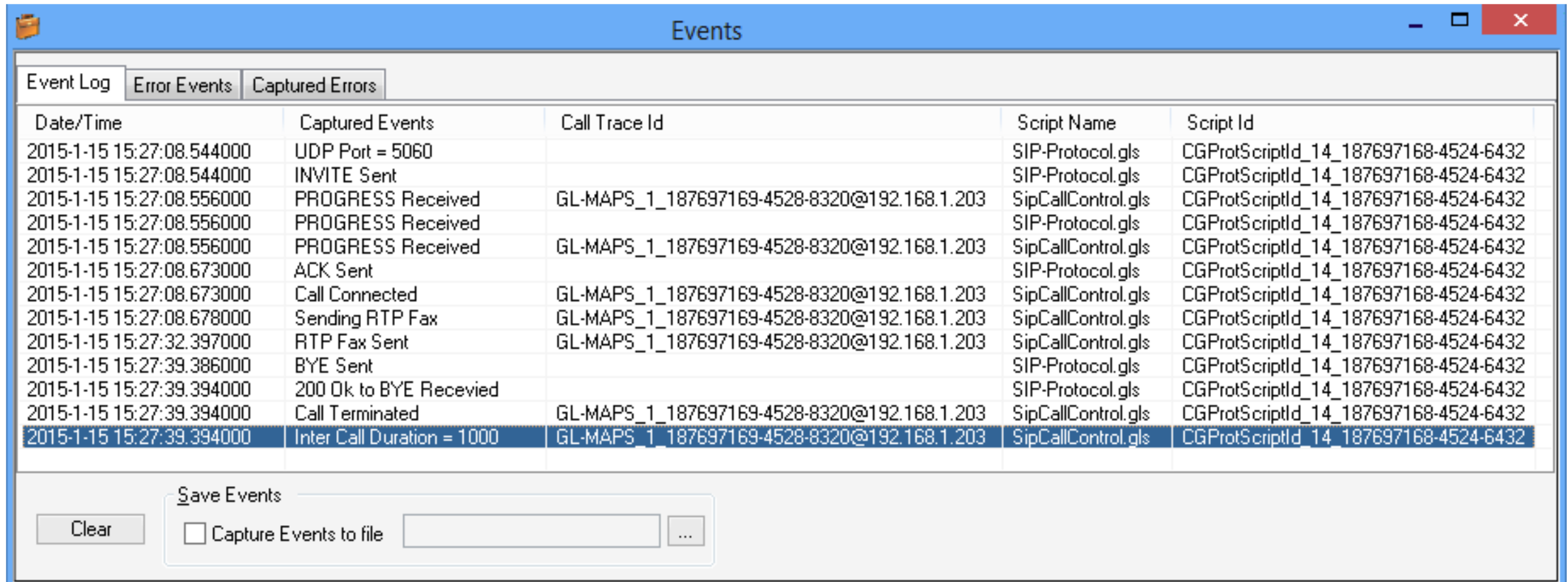
SIP Message Details (Right Window):

```

INVITE sip:0003@192.168.1.143 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.141:5060;branch=z9hG4bK_1_14831002-308-3768
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER
From: 0001 <sip:0003@192.168.1.141>;tag=FromTag_1_14830463-303-3768
To: 0001 <sip:0003@192.168.1.143>
Call-ID: GL-MAPS_1_14830463-305-3768@192.168.1.141
CSeq: 2 INVITE
Contact: 0010 <sip:0003@192.168.1.141>
Content-Type: application/sdp
Content-Length: 359

v=0
o=0001 33852938 33852938 IN IP4 192.168.1.141
s=SIP Call
c=IN IP4 192.168.1.141
t=0
m=image 1028 udpt1 t38
a=T38FaxVersion:3
a=T38MaxBitRate:33600
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:400
a=T38FaxMaxDatagram:280
a=T38FaxUpdEC:t38UDPRedundancy
    
```

FAX Traffic Events



The screenshot shows a window titled "Events" with a tabbed interface. The "Event Log" tab is active, displaying a table of captured events. The table has five columns: Date/Time, Captured Events, Call Trace Id, Script Name, and Script Id. The events are listed in chronological order, starting with a UDP Port = 5060 and ending with an Inter Call Duration = 1000. The last row is highlighted in blue.

Date/Time	Captured Events	Call Trace Id	Script Name	Script Id
2015-1-15 15:27:08.544000	UDP Port = 5060		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.544000	INVITE Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.556000	PROGRESS Received	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.673000	ACK Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.673000	Call Connected	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:08.678000	Sending RTP Fax	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:32.397000	RTP Fax Sent	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.386000	BYE Sent		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	200 Ok to BYE Received		SIP-Protocol.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	Call Terminated	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432
2015-1-15 15:27:39.394000	Inter Call Duration = 1000	GL-MAPS_1_187697169-4528-8320@192.168.1.203	SipCallControl.gls	CGProtScriptId_14_187697168-4524-6432

At the bottom of the window, there is a "Save Events" section with a "Clear" button, a checkbox labeled "Capture Events to file", and a text input field with a browse button ("...").

File Traffic Events

Date/Time	Captured Events	Call Trace Id	Script M
2015-1-15 15:32:20.946000	UDP Port = 5060		SIP-Pro
2015-1-15 15:32:20.946000	INVITE Sent		SIP-Pro
2015-1-15 15:32:20.958000	PROGRESS Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:20.958000	PROGRESS Received		SIP-Pro
2015-1-15 15:32:20.958000	PROGRESS Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.073000	ACK Sent		SIP-Pro
2015-1-15 15:32:21.073000	Call Connected	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.074000	RxFilename = C:\Program Files\GL Communications Inc\MAPS-SIP\VoiceFiles\SIP_9.glw	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.076000	Receiving RTP File	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:21.076000	Sending RTP File	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:41.093000	RTP File Received	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:32:41.093000	RTP File Sent	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:33:21.083000	BYE Sent		SIP-Pro
2015-1-15 15:33:21.091000	200 Ok to BYE Received		SIP-Pro
2015-1-15 15:33:21.091000	Call Terminated	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC
2015-1-15 15:33:21.091000	Inter Call Duration = 1000	GL-MAPS_1_188009580-4552-5476@192.168.1.203	SipCallC

Save Events

Capture Events to file

Video Call Generation

MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Call Generation - CallGenDefault]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result	Total Iter
1	SipCallControl.gls	Profile0001	GL-MAPS_3_851042897-7265-11744@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	
2	SipCallControl.gls	Profile0001	GL-MAPS_3_851045200-7276-5692@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
3	SipCallControl.gls	Profile0001	GL-MAPS_3_851046272-7287-7876@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
4	SipCallControl.gls	Profile0001	GL-MAPS_3_851047176-7298-2364@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
5	SipCallControl.gls	Profile0001	GL-MAPS_3_851048304-7309-11840@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
6	SipCallControl.gls	Profile0001	GL-MAPS_11_851048991-7320-9392@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
7	SipCallControl.gls	Profile0001	GL-MAPS_9_851049784-7327-11744@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
8	SipCallControl.gls	Profile0001	GL-MAPS_9_851050200-7334-5692@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
9	SipCallControl.gls	Profile0001	GL-MAPS_9_851050815-7341-7876@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1
10	SipCallControl.gls	Profile0001	GL-MAPS_9_851052304-7348-2364@192.168.12.78	Stop	Sending Video	SIP_TerminateCall		Pass	1

Add Delete Insert Refresh Start Start All Stop Stop All Abort Abort All

Save Column Width Show Latest

MAPS DUT

```

sequenceDiagram
    participant MAPS
    participant DUT
    MAPS->>DUT: INVITE
    DUT-->>MAPS: 100 Trying
    DUT-->>MAPS: 180 Ringing
    DUT-->>MAPS: 200 OK
    MAPS->>DUT: ACK
    
```

Transmit pre-recorded video traces

Find

```

Content-Type: application/sdp
Content-Length: 291

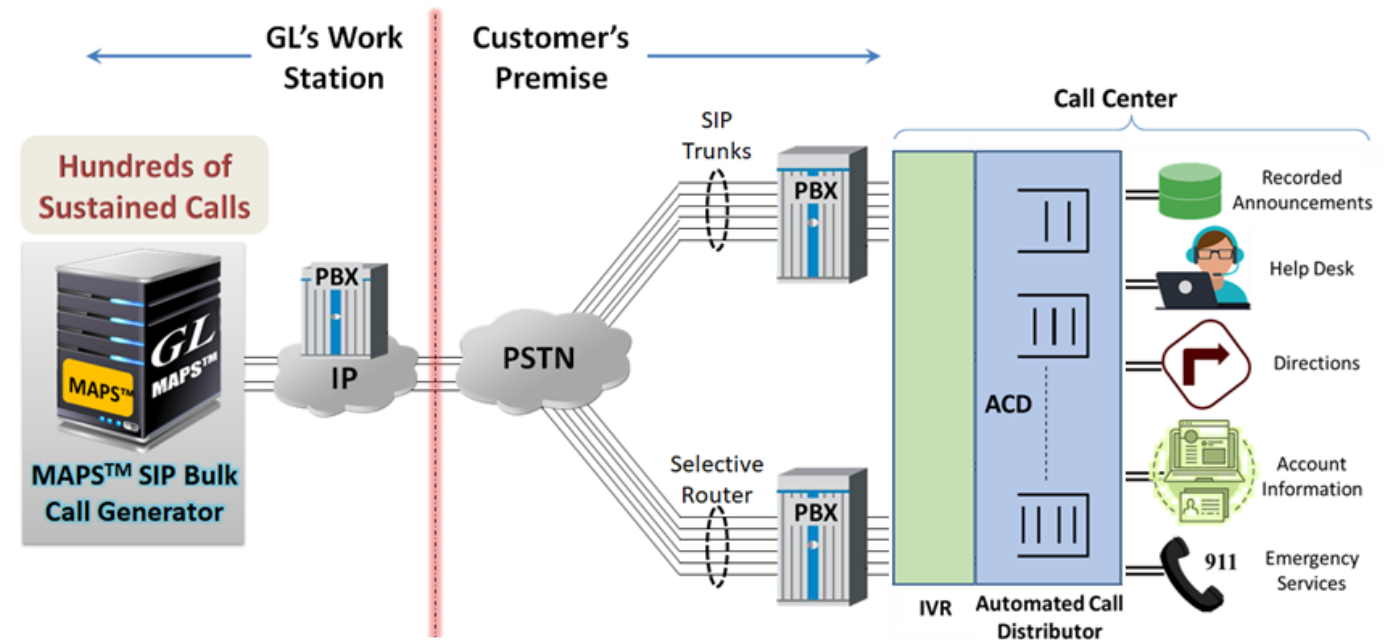
v=0
o=0001 33852938 33852938 IN IP4 192.168.12.74
s=-
c=IN IP4 192.168.12.74
t=0 0
m=audio 1028 RTP/AVP 0
a=rtpmap:0 PCMU/8000
aptime:20
a=sendrecv
m=video 1030 RTP/AVP 97
b=TIAS:256000
a=sendrecv
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42e01e; packetization-mode=1
    
```

Scripts Message Sequence Event Config Script Flow

Initialisation Errors Error Events Captured Errors Link Status Up=0 Down=0

Speech to Text Interactive Voice Response (IVR)

- MAPS™ SIP with GL's Speech Transcription Server provides automated IVR testing by using speech to text to navigate through an IVR tree
- IVR prompts are recorded by MAPS™ SIP and transcribed by the Speech Transcription Server
- Transcribed text is compared to an expected text at each IVR stage to confirm the prompt
- Once the IVR prompt is confirmed, MAPS™ sends DTMF or voice-based responses to move to the next stage
- The expected IVR prompts and responses are defined by the customer to ensure completely customizable tests that are suitable for all IVR systems



GL's Interactive Voice Response Scenario



- The CSV file in the screenshot below shows a basic IVR traversal test of this IVR system

	A	B	C	D	E	F	G
1	IVRIndex	IVRPromptLanguage	IVRExpectedTranscript	IVRResponseType	IVRResponseDTMF	IVRResponseSpeech	IVRNextPromptId
2	int	string	string	string	string	string	int
3	1	en-US	Welcome to GL Communications If you know your partys extension you can dial it at any time For sales press one for technical support press 2 for a directory by last name press 3	DTMF	3		2
4	2	en-US	Welcome to the directory. please enter the first 3 letters of your partys last name using your touch tone keypad Use the seven key for q and the nine key for z	DTMF	926		0

IVR Call Simulation

GL MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) - [Call Generation - CallGenDefault]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Events Profile	Result
3	SipCallControl.gls	Xbox	GL-MAPS-16-1305510-5309-6508@50.76.16.185	Start	PCMU Call Terminated	None		Pa
4	SipCallControl.gls	Profile0001	GL-MAPS-20-59922966-5347-4708@50.76.16.185	Start	PCMU Call Terminated	None		Pa

Column Width Show Latest

Message	Time
ACK	07:26:35.042000
Stage 1: Welcome to GL communications	07:26:41.452000
Stage 1: If you know your parties extension you can download at anytime	07:26:45.625000
Stage 1: For sales press 1	07:26:47.072000
Stage 1: For Technical Support Press 2	07:26:49.996000
Stage 1: Or directory by last name press 3	07:26:53.245000
Digits Transmitted :: 3	07:26:53.504000
Stage 2: Welcome to the directory Please enter the first 3 letters of your party's last name	07:27:06.247000
Stage 2: Using your touch tone keypad use the Seven key for Q and the nine key for Z I'm sorry	07:27:19.124000
Digits Transmitted :: 926	07:27:19.826000
BYE	07:27:20.786000
200 OK	07:27:20.949000

```

INVITE sip:13016704784@104.219.163.73 SIP/2.0
Via: SIP/2.0/UDP 50.76.16.185:5060;branch=z9hG4bK-21-59922966-5348-4708
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER, UPDATE
From: 3016704784 <sip:3016704784@50.76.16.185>;tag=FromTag-18-59922966-5345-4708
To: 13016704784 <sip:13016704784@104.219.163.73>
Call-ID: GL-MAPS-20-59922966-5347-4708@50.76.16.185
CSeq: 1 INVITE
Contact: 3016704784 <sip:3016704784@50.76.16.185>
Content-Type: application/sdp
Content-Length: 242

v=0
o=3016704784 38929794 1 IN IP4 50.76.16.185
s=SIP Call
c=IN IP4 50.76.16.185
t=0 0
m=audio 1026 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
aptime:20
a=sendrecv
  
```

Initialisation Errors
 Error Events
 Captured Errors
 Link Status Up=0 Down=0

IVR Call Simulation Reports

SIP IVR Detailed Log

Maps_IVR_DetailedLog_2020-05-05_08-25-32_Profile0001.pdf - Adobe Acrobat Reader DC

File Edit View Window Help

Home Tools Maps_IVR_DetailedLog_2020-05-05_08-25-32_Profile0001.pdf x

59.1%

GL Communications Inc Date: 05/05/2020

MAPS IVR Test Start Time: 08:25:32

Destination number: 13016704784

Time	Type	Event	Certainty	Stage	Received Prompt	Expected Prompt	Similarity
2020-05-05 08:25:39.088000	Rx	Welcome to GL communications	0.8831	1			
2020-05-05 08:25:39.087000	Analysis			1	Welcome to GL communications	Welcome to GL Communications If you know your partys extension you can dial it at any time For Sales press one for technical support press 2 for a directory by last name press 3	15.819208
2020-05-05 08:25:42.775000	Rx	If you know your parties extension you can download at anytime	0.9424	1			
2020-05-05 08:25:42.775000	Analysis			1	Welcome to GL communications If you know your partys extension you can download at anytime	Welcome to GL Communications If you know your partys extension you can dial it at any time For Sales press one for technical	45.762711
2020-05-05 08:25:44.468000	Rx	For sales press 1	0.8577	1			
2020-05-05 08:25:44.469000	Analysis			1			
2020-05-05 08:25:51.230000	Rx	For Technical Support Press 2 for directory by last name press 3	0.9056	1			
2020-05-05 08:25:51.230000	Analysis			1			
2020-05-05 08:25:51.231000	Tx	3					
2020-05-05 08:25:52.511000	Rx	For a directory by First Name Press	0.8785	1			
2020-05-05 08:26:21.479000		Failed to transcribe audio		2			

SIP IVR Result Log

MAPS_SIP_IVR_Result_2020_05_05_08_25_25.pdf - Adobe Acrobat Reader DC

File Edit View Window Help

Home Tools MAPS_SIP_IVR_Res... x

59.1%

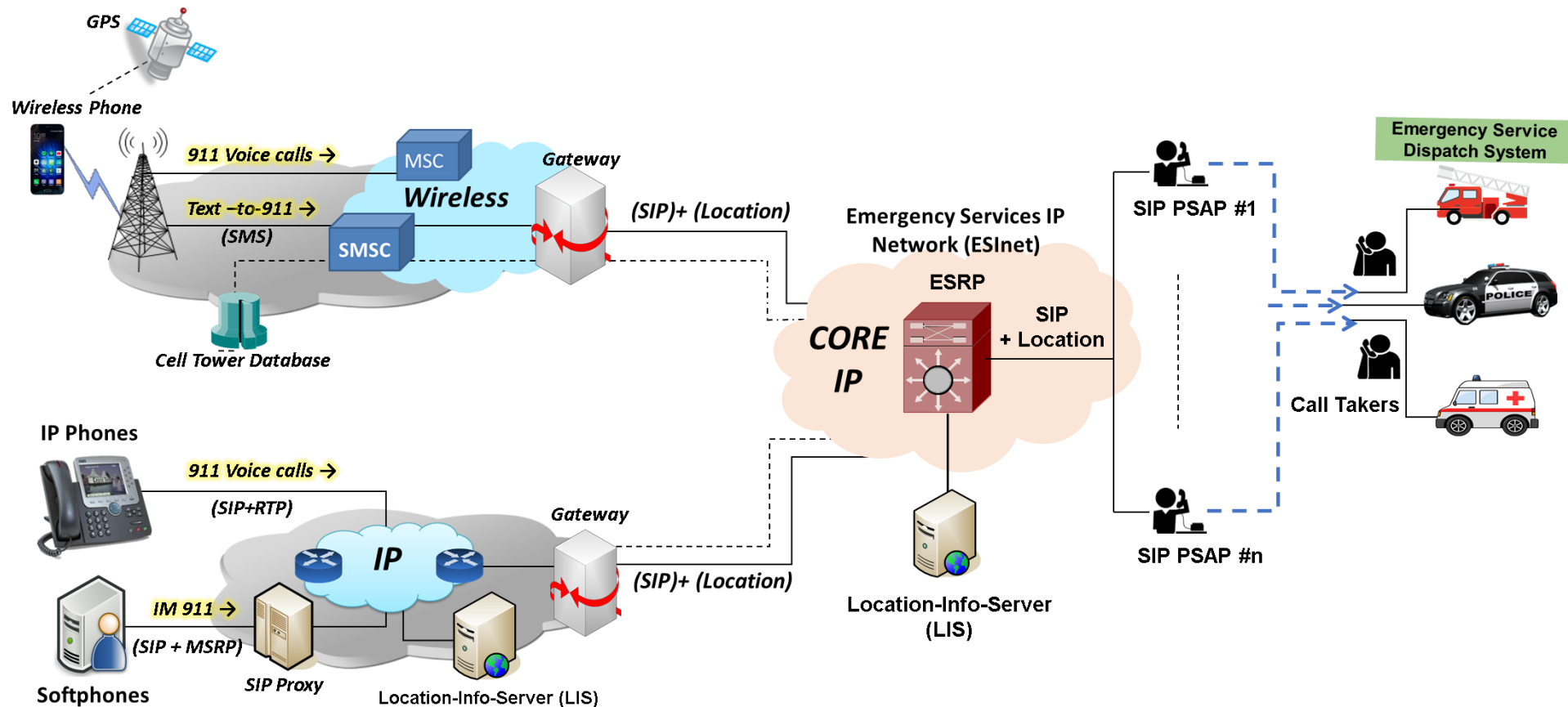
GL Communications Inc Date: 05/05/2020

MAPS SIP IVR Test Start Time: 08:25:25

SI.No	Time	Profile	Destination TN	IVR File	SIP Result	IVR Result	Detailed Report
1	2020-05-05 08:26:22.979000	Profile0001	13016704784	maps/sip/ietf/ivr_prompt_gl.csv	Pass	Pass	MAPS/SIP/ET/IVR/Log/De tailedLog/Maps_IVR_Detailed Log_2020-05-05_08:25- 32_Profile0001.pdf

Message Session Relay Protocol

Message Session Relay Protocol is a text-based, connection-oriented protocol for transmitting a series of related instant messages in the context of a session. MSRP sessions are typically arranged using SIP the same way a session of audio or video media is set up.



Message Session Relay Protocol (Contd.)

- Reads text messages from a pre-defined text file (user-configurable) and transmits them on established IM session
- Received messages on every MSRP session can be recorded to a text file
- Text file can have multiple lines of message. The CRLF will be the de-limiter to treat each line as a new message
- Supports message chunking with user configured chunk size
- Configuration options allow to:
 - Record and report success and failure reports in MSRP SEND method
 - Define message generation interval to control the message frequency on the call
- Supports mixed media SIP sessions i.e., Audio with IM / Video with IM / Only IM
- Provides IM statistics per call and aggregated statistics of over-all calls (Number and size of messages received and sent)
- Flexibility to validate MSRP devices through negative tests with invalid MSRP URI's, validate success and failure reports
- Supports up to 500 simultaneous MSRP sessions

MSRP Traffic Configuration

The screenshot displays the MAPS (Message Automation Protocol Simulation) software interface, specifically the Profile Editor for TrafficProfile. The main window is titled "GL MAPS (Message Automation Protocol Simulation) (SIP IETF) - [Profile Editor - TrafficProfile]". The interface includes a menu bar (Configurations, Emulator, Reports, Editor, Debug Tools, Windows, Help) and a toolbar with various icons.

The main area is divided into two panes. The left pane, titled "Profiles (Edt-F2)", lists ten profiles, with "Profile0001" selected. The right pane, titled "Config", shows a tree view of configuration options. The "MSRP Text Message Configurations" folder is expanded, and the "Send IM" sub-folder is selected. The "Send IM" configuration is displayed in a table:

Config	Value
Send Recv T38 Fax	
Tx T38 Fax File Name	C:\Program Files\GL Communications Inc\MAPS-SIP\Fa...
T38 Rx Fax Path	C:\Program Files\GL Communications Inc\MAPS-SIP\Fa...
T38 Rx Fax File Prefix	SIP
Rx File Creation Type	Random Number
TxVideo	
RTP Transport Type	UDP
Video Trace File Path	videofiles\pcmu-h264.hdl
Mute Audio RTP Stream	Disable
Mute Video RTP Stream	Disable
MSRP Text Message Configurations	
Send IM	
IM File Name	imfiles\send\msrpinputmessage.txt
IM File Iterations	1
Inter IM Timeout in msec	1000
IM Chunking Size	0
IM Success Report	no
IM Failure Report	yes
Recv IM	
Rx IM File Path	C:\Program Files\GL Communications Inc\MAPS-SIP\I...
Rx IM File Creation Type	Sequence Number
Rx IM File Prefix	SIP-IM

The "IM File Name" row is highlighted with a blue border. To the right of the configuration pane, a "Select File" dialog box is open, showing the file path "imfiles\send\msrpinputmessage.txt" and an "Open" button. Below the dialog, a Notepad window titled "MsrpInputMessage.txt - Notepad" is open, displaying the following text:

```
File Edit Format View Help
Hi, Welcome
This is MAPS SIP MSRP Simulator.
Test Message 1.
Test Message 2.
Test Message 3.
```

The Notepad window also shows "Add", "Insert", and "Delete" buttons, and a "Properties" button. At the bottom of the MAPS interface, there are status indicators for "Initialisation Errors", "Error Events", "Captured Errors", and "Link Status Up=0 Down=0".

MSRP Call Generation

MAPS (Message Automation Protocol Simulation) (SIP) (MSRP) - [Call Generation - BulkCalls_10]

Configurations Emulator Reports Editor Debug Tools Windows Help

Sr No	Script Name	Profile	Call Info	Script Execution	Status	Events	Ev	Result	Total Iterations	Completed Iterations
1	SipCallControl.gls	Profile0001	GL-MAPS_457_86849705-8370-17280@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
2	SipCallControl.gls	Profile0002	GL-MAPS_458_86849705-8374-14176@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
3	SipCallControl.gls	Profile0003	GL-MAPS_458_86849705-8366-2664@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
4	SipCallControl.gls	Profile0004	GL-MAPS_468_86849705-8358-4812@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
5	SipCallControl.gls	Profile0005	GL-MAPS_470_86849705-8363-17328@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
6	SipCallControl.gls	Profile0006	GL-MAPS_467_86849704-8354-16532@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
7	SipCallControl.gls	Profile0007	GL-MAPS_462_86849706-8386-17280@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
8	SipCallControl.gls	Profile0008	GL-MAPS_463_86849707-8394-14176@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
9	SipCallControl.gls	Profile0009	GL-MAPS_463_86849706-8390-2664@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0
10	SipCallControl.gls	Profile0010	GL-MAPS_473_86849706-8381-4812@192.168.12.216	Stop	Call Connected	SIP_TerminateCall		Pass	1	0

Add Delete Insert Refresh Start Start All Stop Stop All Abort Abort All

Save Column Width Show Latest

MAPS DUT

```

INVITE → 15:39:35.705000
← 100 Trying 15:39:35.727000
← 180 Ringing 15:39:35.737000
← 200 OK 15:39:35.859000
ACK → 15:39:35.861000
SEND → 15:39:35.909000
← 200 OK 15:39:35.949000
REPORT → 15:39:35.991000
← SEND 15:39:35.991000
← 200 OK 15:39:35.992000
REPORT → 15:39:36.010000
← SEND 15:39:36.943000
← 200 OK 15:39:36.998000
REPORT → 15:39:37.030000
← SEND 15:39:37.030000
← 200 OK 15:39:37.032000
REPORT → 15:39:37.040000
                    
```

Find

```

MSRP glMapsMsrpBB9A66F9-153935908-6777 SEND
To-Path: msrp://192.168.12.209:20148/GL_MAPS_302_86849888;tcp
From-Path: msrp://192.168.12.216:20151/GL_MAPS_464_86849744;tcp
Message-ID: glMapsMsrpBB9A66F9-153935908-6776
Success-Report: no
Failure-Report: yes
Byte-Range: 1-270/270
Content-Type: text/plain

GL's Message Automation & Protocol Simulation (MAPS™) is a protocol simulation and conformance test tool that supports a variety of
-----glMapsMsrpBB9A66F9-153935908-6777?
                    
```

Scripts Message Sequence Event Config Script Flow

MSRP Statistics

Name	Values
*****... ..	0
Total MSRP Messages Sent	340
Total MSRP Messages Received	345
Total MSRP Message Bytes Sent	15285
Total MSRP Message Bytes Received	15285

Load Generation

Load Generation - LoadGendefault

Total Calls To Generate: * (* indicates no limit)
Max Active Calls: 2000 Unique Distributions Per Script

Multi Distributions

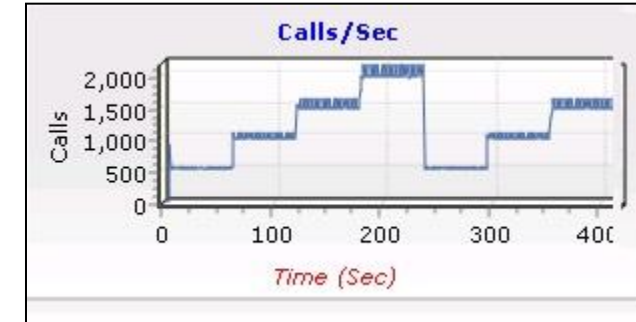
Distributions	Description	
Uniform	MinCR=40 , MaxCR=80 , Duration=10	Add
Fixed	Call Rate=250 , Duration=10	Remove
Normal	MinCR=40 , MaxCR=80 , Duration=10	Remove All
		Edit

Scripts Exclusive Profiles

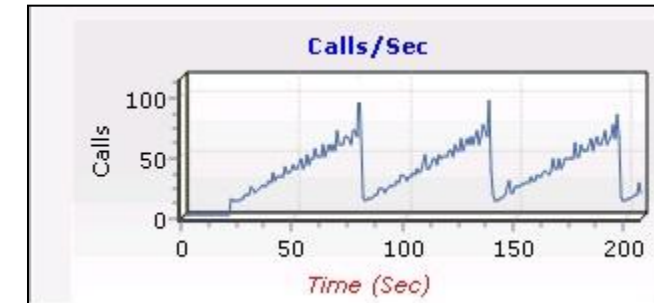
Scripts	Profile
SipCallControl	Profile0003
Registration	Profile0005

Stop Time
Days: 0 Hours: 0 Minutes: 0
Start Time - 00:00:00.000 End Time - 00:00:00.000
Pause Start

Step Statistical Distribution



Ramp Statistical Distribution

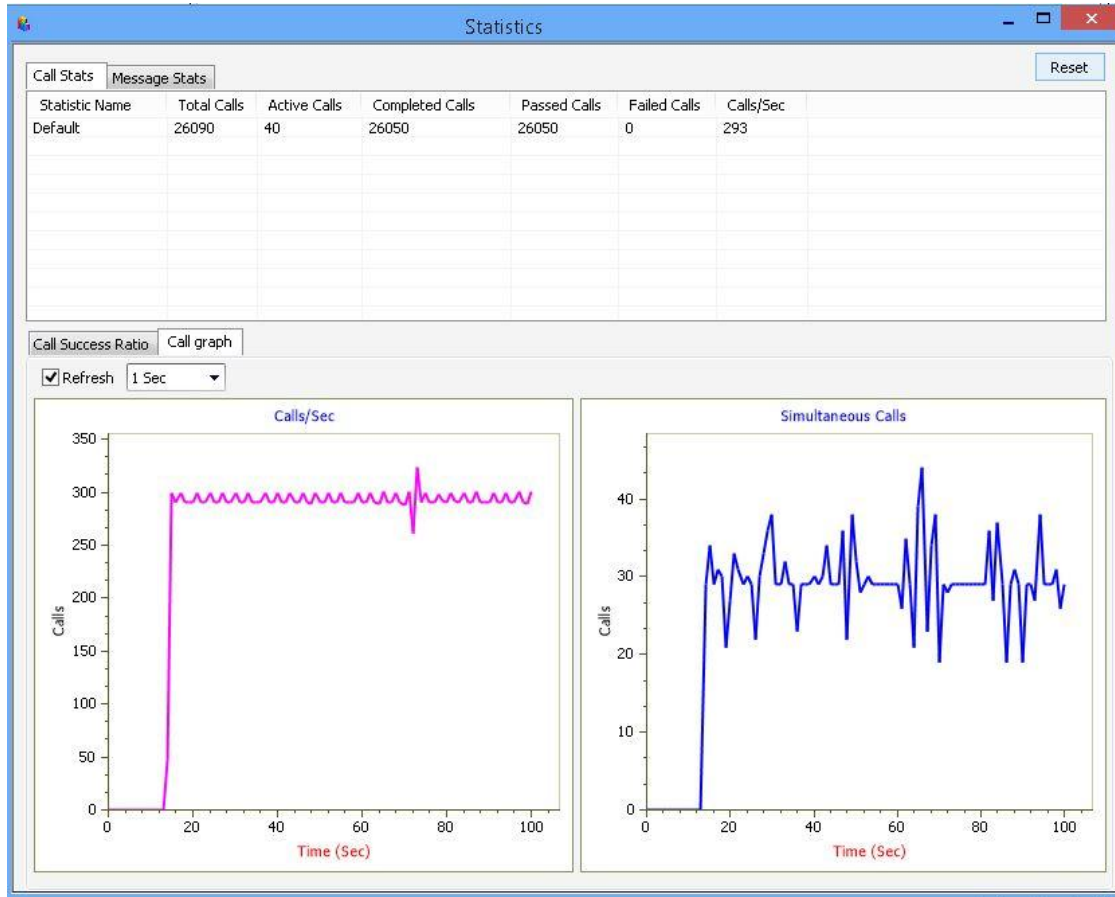


Saw-tooth Statistical Distribution

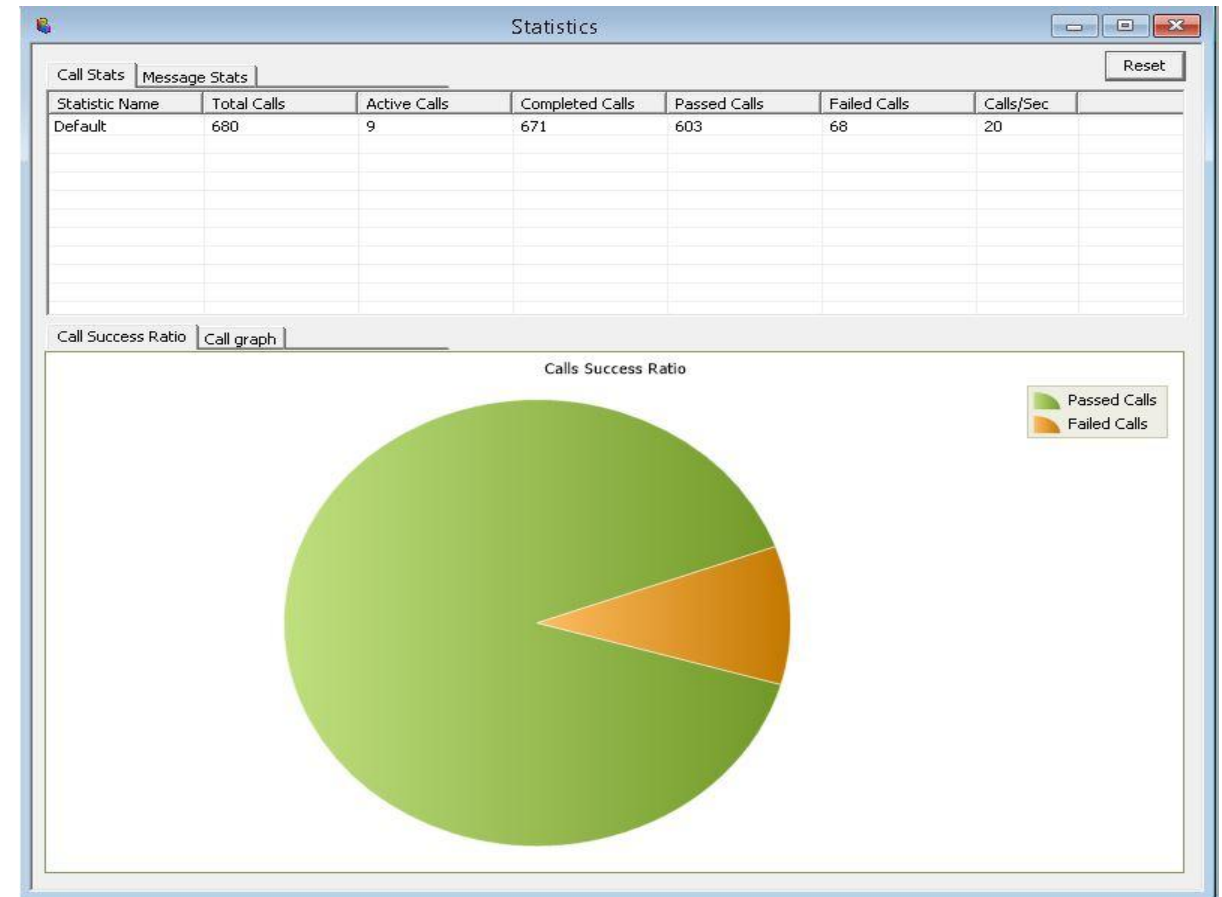


Success Call Ratio Statistics

Call Graph



Call Stats



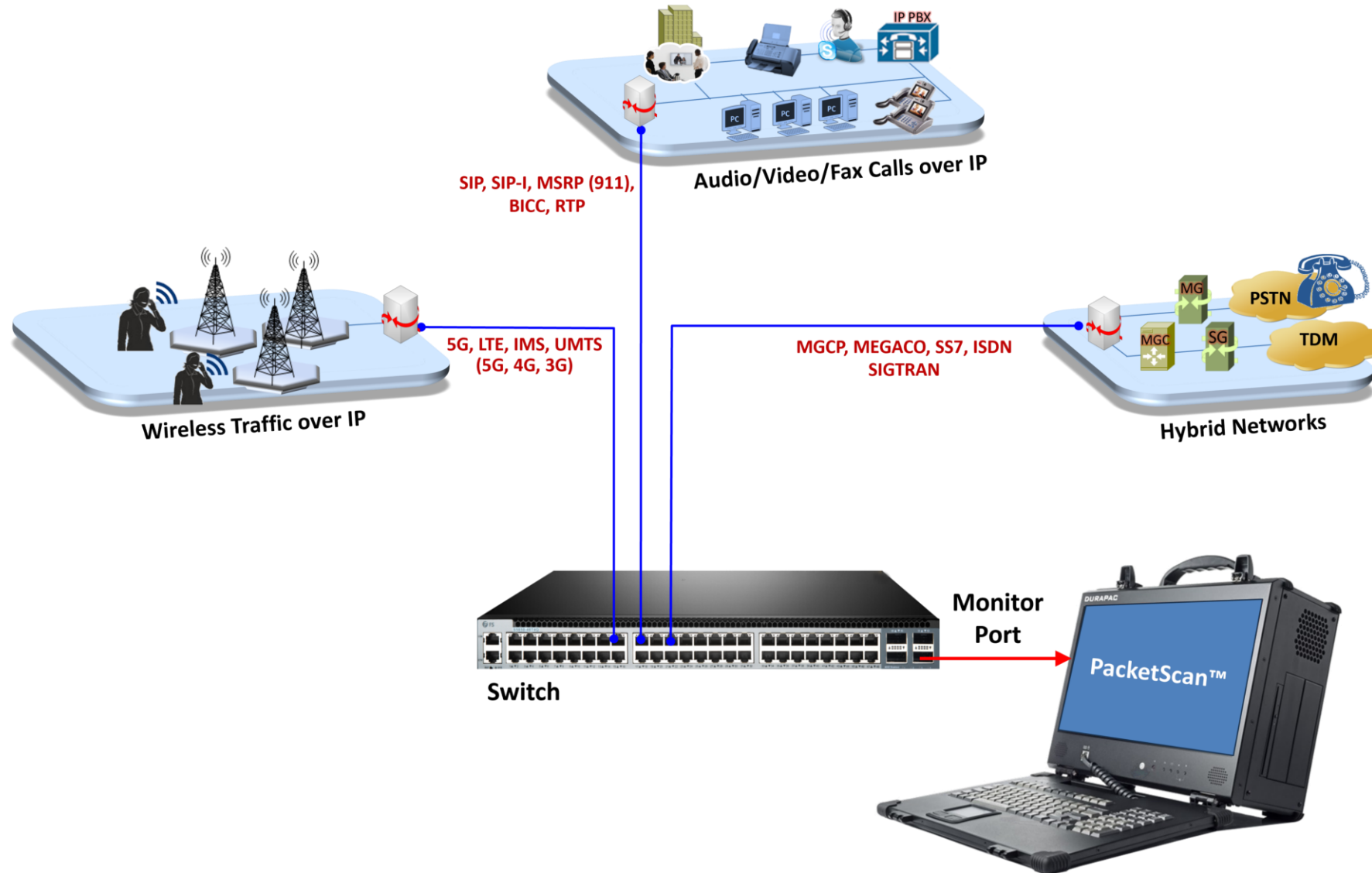
Message Statistics

Message Type	Tx Count	Rx Count	Retransmit Count
100 INVITE	0	66040	0
180 INVITE	0	66040	0
200 BYE	0	46808	0
200 INVITE	0	66040	0
ACK	66040	0	0
BYE	46808	0	0
INVITE	66040	0	0

SIP RTP Analyzer - PacketScan™

PacketScan™ VoIP Traffic Analysis

SIP / MSRP / H323 / MEGACO / MGCP / RTP / RTCP / Video Analysis



SIP Decode in PacketScan™

The screenshot shows the PacketScan (All-in-One) interface. The top menu includes File, View, Capture, Statistics, Database, Call Detail Records, Configure, and Help. Below the menu is a toolbar with various icons and a 'GoTo' field containing the number '0'. The main window displays a table of captured packets:

Dev	Frame#	TIME (Relative)	Len	Error	Protocols	IP Packet Type	Source IP Address	Destination IP Address	UDP Source Port	UDP Destination Port
✓ 2	0	00:00:00.000000	836		Internet IP(IPv4)	SIP	192.168.1.200	192.168.1.103	54098	5060
✓ 2	1	00:00:00.001552	354		Internet IP(IPv4)	SIP	192.168.1.103	192.168.1.200	54098	5060
✓ 2	2	00:00:00.001669	355		Internet IP(IPv4)	SIP	192.168.1.103	192.168.1.200	54098	5060
✓ 2	3	00:00:04.487598	820		Internet IP(IPv4)	SIP	192.168.1.103	192.168.1.200	54098	5060

Below the table, the detailed decode for frame 0 is shown, titled 'Sip3261 Layer':

```

===== Sip3261 Layer =====
HDR      = INVITE sip:0001@192.168.1.103 SIP/2.0
HDR      = Via: SIP/2.0/UDP 192.168.1.200:5060;branch=z9hG4bK3811333536-3
HDR      = Max-Forwards: 70
HDR      = Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, COMET, OPTIONS, SUBSCRIB
HDR      = From: 0001 <sip:0001@192.168.1.200>;tag=GLPG_3811333536-333
HDR      = To: 0001 <sip:0001@192.168.1.103>
HDR      = Call-ID: GLPG-483633760331
HDR      = CSeq: 1 INVITE
HDR      = Contact: 0001 <sip:0001@192.168.1.103>
HDR      = Content-Type: application/sdp
HDR      = Content-Length: 349
BODY     = v=0
BODY     = o=0001 47706128 47706129 IN IP4 192.168.1.200
BODY     = s=-
BODY     = c=IN IP4 192.168.1.200
BODY     = t=0 0
BODY     = m=audio 1024 RTP/AVP 0 8 18 104 3 101
BODY     = a=rtpmap:0 PCMU/8000/1
BODY     = a=rtpmap:8 PCMA/8000/1
BODY     = a=rtpmap:18 G729/8000/1
BODY     = a=fmtp:18 annexb=no
BODY     = a=rtpmap:104 G726-32/8000/1
BODY     = a=rtpmap:3 GSM/8000/1
BODY     = a=rtpmap:101 telephone-event/8000
BODY     = a=fmtp:101 0-15
BODY     = a=ptime:20
BODY     = a=sendrecv
    
```

The status bar at the bottom indicates 'Off-line Viewing' and 'C:\Program Files\GL Communications Inc\P 2 550 Frames'.

PacketScan™ PDA with SIP Call Summary

Traffic Analyzer - Summary View

File View Call Summary Settings Help

Sip Calls Show All Sessions

Call Summary Registraton Summary Alert Summary

Call #	SSRC	Payload	Packet Received	Conversational MOS/R-Factor	Listening MOS/R-Factor	Packets Discard...	Missing Packets...	Duplicate Packets...	Out Of Sequen...	Average Gap[ms]	Average Delay	Average Jitter	Average Inter Arri...	Cumulativ Packet ...	Max/Min Gap	Max/Min Delay	Max/M Jitter
Call#000001 Caller:0001@192.168.1.203 Callee:0001@192.168.1.213 CallId:GL-MAPS_1_185372727-4480-8320@192.168.1.203 Call StartTime:2015-01-15 14:48:24.106 Call Duration: 00:01:00.023																	
1	2217509121	PCMU/8000	1005	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	21.17 ...	1 / -1	0.45 /
1	2217326337	PCMU/8000	146	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	20.09 ...	0 / 0	0.07 /

TimeStamp	192.168.1.203	192.168.1.213
00.00.000	5060	5060
		INVITE
00.00.007	5060	5060
		SIP/2.0 100 Trying
00.00.009	5060	5060
		SIP/2.0 180 Ringing
00.00.132	5060	5060
		SIP/2.0 200 OK
00.00.137	5060	5060
		ACK
00.00.141	1036	1036
		RTP (PCMU/8000)
00.00.147	1036	1036
		RTP (PCMU/8000)
01.00.156	5060	5060
		BYE
01.00.160	5060	5060
		SIP/2.0 200 OK

```

INVITE sip:0001@192.168.1.213 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.203:5060;branch=z9hG4bK_1_185372727-4481-8320
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, PRACK, OPTIONS, NOTIFY, REGISTER, UPDATE
From: "MapsSip" <sip:0001@192.168.1.203>;tag=FromTag_1_185372727-4478-8320
To: 0001 <sip:0001@192.168.1.213>
Call-ID: GL-MAPS_1_185372727-4480-8320@192.168.1.203
CSeq: 1 INVITE
Contact: 0010 <sip:0001@192.168.1.203>
Content-Type: application/sdp
Content-Length: 317

v=0
o=0001 33852938 33852938 IN IP4 192.168.1.203
s=-SIP Call
c=IN IP4 192.168.1.203
t=0 0
m=audio 1036 RTP/AVP 0 8 18 3 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:3 GSM/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
aptime:20
    
```

Active Calls Graph Average Jitter Distribution E-Model RTP Packets Graph T.38 Analysis Call Graph Call Summary

PacketScan™ Fax T.38 Analysis

Traffic Analyzer - Summary View

File View Call Summary Settings Help

Sip Calls Show Fax Calls

Call Summary Registrar Summary Alert Summary

Call #	SSRC	Payload	Packet Received	Conversational MOS/R-Factor	Listening MOS/R-Fac...	Packets Discard...	Missing Packets...	Duplicate Packets...	Out Of Sequen...	Average Gap(ms)	Average Delay	Average Jitter	Average Inter Arri...	Cumulativ Packet ...	Max/Min Gap	Max/Min Delay	Max/Min Jitter	Max RTT
F Call#000001 Caller:4000@192.168.1.60 Callee:1000@192.168.1.60 CallId:1620788079-5060-3@192.168.1.244 Call StartTime:2011-09-02 12:35:48.113 Call Duration: 00:02:39.529																		
1	390089559	PCMU/8000	698	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	20.10 ...	0 / 0	0.08 / ...	0.0
1	1321168996	PCMU/8000	697	4.20 / 93	4.20 / 93	0 / 0.00	0 / 0.00	0 / 0.00	0 / 0.00	20.00	0.00	0.00	0	0	20.07 ...	0 / 0	0.08 / ...	0.0

Time	SSRC	Direction	Payload	SSRC
00.30.538	5004	←	v21-preamble	5004
00.31.580	5004	←	NSF	5004
00.31.955	5004	←	CSI NUM:918040488401	5004
00.32.648	5004	←	DIS:DSR:ITU-T V.27 ter and V.29	5004
00.33.110	5004	←	no-signal	5004
00.34.559	5004	→	v21-preamble	5004
00.35.657	5004	→	TSI NUM:40488401	5004
00.36.402	5004	→	DCS:DSR:9600bps, ITU-T V.29	5004
00.36.622	5004	→	no-signal	5004
00.36.914	5004	→	v29-9600-training	5004
00.37.156	5004	→	t4-non-ecm-data:v29-9600: 0 pkts lost	5004
00.38.678	5004	→	no-signal	5004

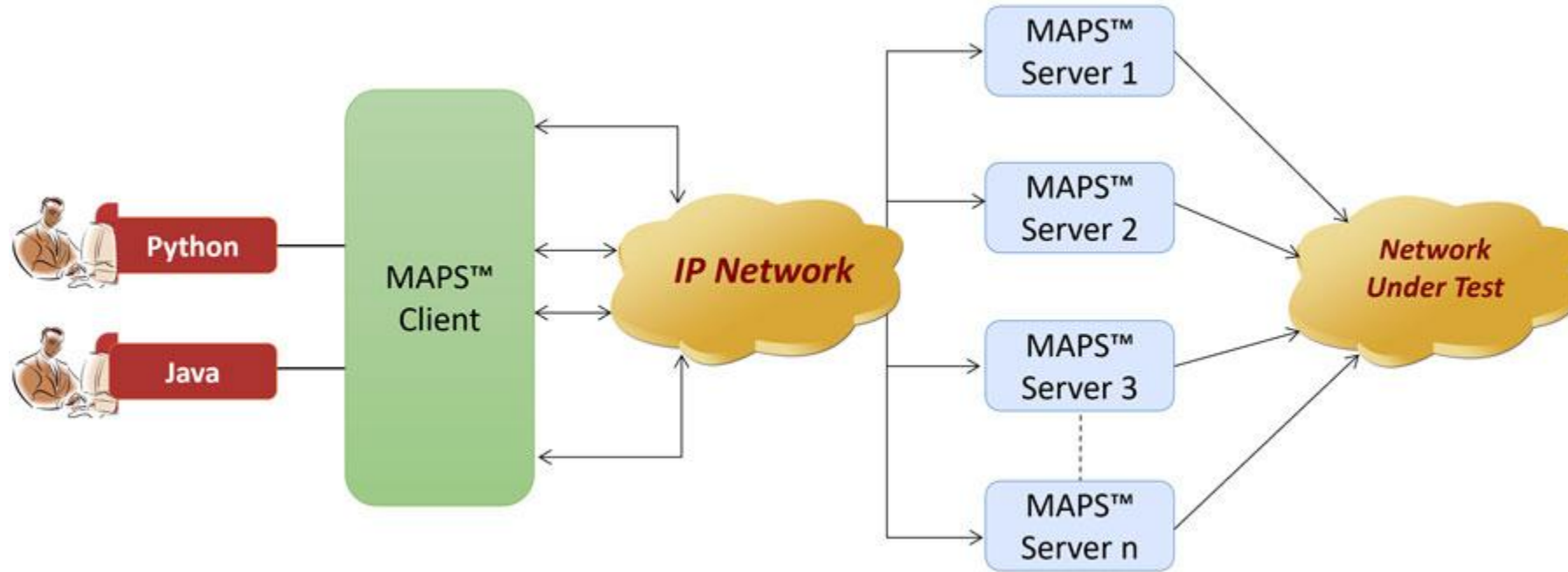

```

===== T.38 Layer =====
UDPTLPacket = SEQUENCE
seq-number = INTEGER
Contents = 6
primary-ifp-packet = Open Type
Length = 12
IFPPacket = SEQUENCE
Preamble = 1
type-of-msg = CHOICE
Choice Index = 1
data = ENUMERATOR
Extensibility Marker = 0
Contents = 0 v21(0)
data-field = SEQUENCE OF
Iteration Count = 2
data-field = Instance 0
data-field = SEQUENCE
Preamble = 1
field-type = ENUMERATOR
Contents = 0 hdlc-data(0)
field-data = OCTET STRING
Length Determinant = 6
Contents = xFFC0042A20EB
data-field = Instance 1
data-field = SEQUENCE
Preamble = 0
field-type = ENUMERATOR

```

Active Calls Graph Average Jitter Distribution E-Model RTP Packets Graph T.38 Analysis Call Graph Call Summary

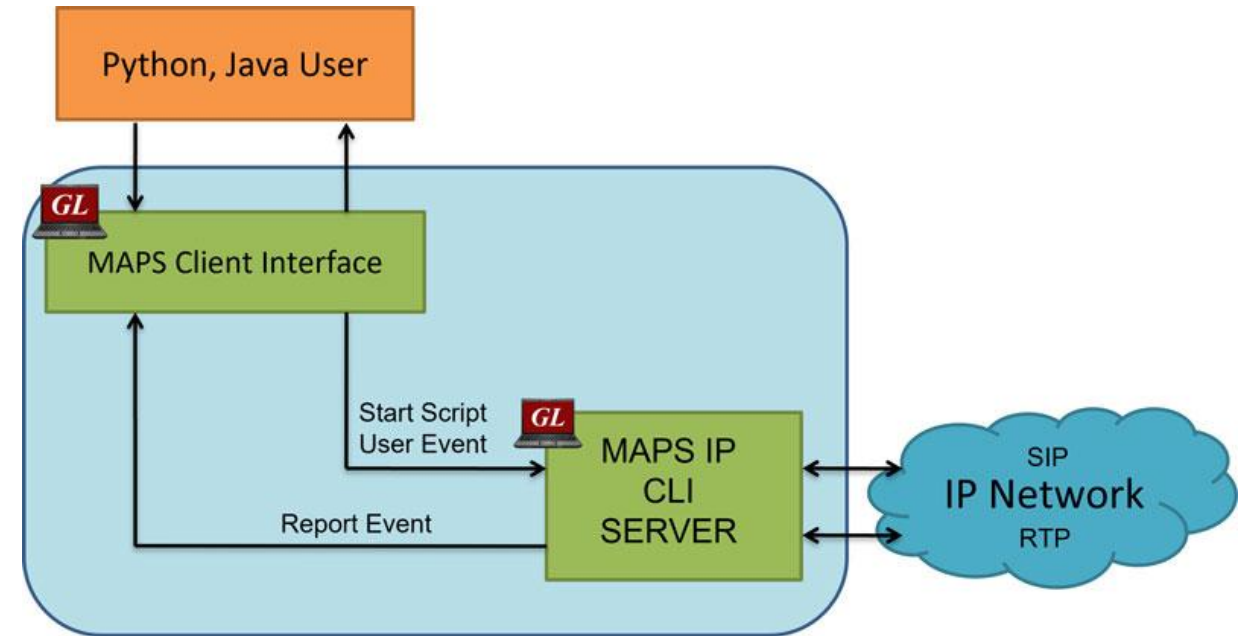
MAPS™ Command Line Interface



- MAPS™ can be configured as server-side application, to enable remote controlling through multiple command-line based clients. Supported clients include Python and Java
- The MAPS™ APIs allows for programmatic and automated control over all MAPS™ platforms. Each MAPS™ server can receive multiple client connections and offer independent execution to each client
- Likewise, a single client can connect to multiple MAPS™ servers, including servers running different protocols, permitting complex cross-protocol test cases

MAPS™ SIP CLI Test System

- As depicted, MAPS™ SIP CLI test system consists of the following -
 - Python, Java user communicating over TCP/IP
 - MAPS™ Client IFC, and MAPS™ SIP CLI Server



MAPS™ CLI Server and Python Client

```
CLI MapsCLI (SIP IETF)
File Edit View
View Latest Command
5 :: 2020-7-3 13:06:18.333000 : Start "TestBedDefault.xml";
5 :: 2020-7-3 13:06:18.442000 : LoadProfile "UserAgent_Profiles.xml"
5 :: 2020-7-3 13:06:18.770000 : Apply Global Configuration # "_EnableCLI"=1;
5 :: 2020-7-3 13:06:18.771000 : StartScript 1 "SipCallControl.gls" "Profile0001" 1;
5 :: 2020-7-3 13:06:18.880000 : UserEvent 1 "SetVariable"# "Contact"="1231230001@192.168.12.216";
5 :: 2020-7-3 13:06:18.991000 : UserEvent 1 "SetVariable"# "AddressOfRecord"="1231230001@192.168.12.216";
5 :: 2020-7-3 13:06:19.105000 : UserEvent 1 "SetVariable"# "RtpIpAddress"="192.168.12.216";
5 :: 2020-7-3 13:06:19.209000 : UserEvent 1 "SetVariable"# "To"="0001@192.168.12.209";
5 :: 2020-7-3 13:06:19.318000 : UserEvent 1 "SetVariable"# "PacketizationTime"="20";
5 :: 2020-7-3 13:06:19.429000 : UserEvent 1 "SetVariable"# "OvrCodecListSize"=3;
5 :: 2020-7-3 13:06:19.540000 : UserEvent 1 "SetVariable"# "OvrCodecList[0]"="G729";
5 :: 2020-7-3 13:06:19.646000 : UserEvent 1 "SetVariable"# "OvrPayloadList[0]"=18;
5 :: 2020-7-3 13:06:19.756000 : UserEvent 1 "SetVariable"# "OvrCodecList[1]"="PCMU";
5 :: 2020-7-3 13:06:19.864000 : UserEvent 1 "SetVariable"# "OvrPayloadList[1]"=0;
5 :: 2020-7-3 13:06:19.979000 : UserEvent 1 "SetVariable"# "OvrCodecList[2]"="telephone-event";
5 :: 2020-7-3 13:06:20.085000 : UserEvent 1 "SetVariable"# "OvrPayloadList[2]"=101;
5 :: 2020-7-3 13:06:20.192000 : UserEvent 1 "RTP_CreateSession";
5 :: 2020-7-3 13:06:24.349000 : UserEvent 1 "GetCallStatus";
5 :: 2020-7-3 13:06:24.460000 : UserEvent 1 "GetNegotiatedCodec";
5 :: 2020-7-3 13:06:24.569000 : UserEvent 1 "SendFile"# "TxFileName"="voicefiles\Send\G711\ULAW\Wijay.glw", "TxFileDuration"=10;
5 :: 2020-7-3 13:06:34.635000 : UserEvent 1 "GetVoiceQualityStats";
5 :: 2020-7-3 13:06:34.739000 : UserEvent 1 "SIP_TerminateCall";
5 :: 2020-7-3 13:06:34.848000 : UserEvent 1 "GetMessageCount";
5 :: 2020-7-3 13:06:34.957000 : UserEvent 1 "GetMessageInfo"# "Index"=0;
5 :: 2020-7-3 13:06:35.067000 : UserEvent 1 "GetMessageInfo"# "Index"=1;
5 :: 2020-7-3 13:06:35.178000 : UserEvent 1 "GetMessageInfo"# "Index"=2;
5 :: 2020-7-3 13:06:35.290000 : UserEvent 1 "GetMessageInfo"# "Index"=3;
5 :: 2020-7-3 13:06:35.397000 : UserEvent 1 "GetMessageInfo"# "Index"=4;
5 :: 2020-7-3 13:06:35.510000 : UserEvent 1 "GetMessageInfo"# "Index"=5;
5 :: 2020-7-3 13:06:35.616000 : UserEvent 1 "GetMessageInfo"# "Index"=6;
5 :: 2020-7-3 13:06:35.724000 : StopScript 1;
ServerLog:errCode = 0, errString = connection has been gracefully closed for ClientId =5
```

```
Python 3.7.3 Shell
File Edit Shell Debug Options Window Help
Python 3.7.3 (v3.7.3:ef4ec6ed12, Mar 25 2019, 22:22:05) [MSC v.1916 64 bit (AMD64)] on win32
Type "help", "copyright", "credits" or "license()" for more information.
>>>
RESTART: C:\Program Files\GL Communications Inc\MAPS-SIP\PythonClient\examples\SIP\SipBasicCall.py
SERVER INITIALIZED
CONNECTED
Negotiated Codec = PCMU
0
CMOS = 4.19531
LMOS = 4.19531
CR_FACTOR = 93
LR_FACTOR = 93
TX_PACKETS = 501
RX_PACKETS = 712
LOST_PACKETS = 0
DISCARDED_PACKETS = 0
OUT_OF_SEQ_PACKETS = 0
DUPLICATE_PACKETS = 0
AVG_JITTER = 0.125

12:24:01.120 -> INVITE
INVITE sip:0001@192.168.12.209 SIP/2.0
Via: SIP/2.0/UDP 192.168.12.216:5060;branch=z9hG4bK-4-1348328288-22704-17372
Max-Forwards: 70
Allow: INVITE, BYE, CANCEL, ACK, INFO, OPTIONS, SUBSCRIBE, NOTIFY, REFER, REGISTER, UPDATE
From: 1231230001 <sip:1231230001@192.168.12.216>;tag=FromTag-1-1348328288-22701-17372
To: 0001 <sip:0001@192.168.12.209>
Call-ID: GL-MAPS-3-1348328288-22703-17372@192.168.12.216
CSeq: 1 INVITE
Contact: 1231230001 <sip:1231230001@192.168.12.216>
Content-Type: application/sdp
Content-Length: 269

v=0
o=1231230001 39377840 1 IN IP4 192.168.12.216
s=SIP Call
c=IN IP4 192.168.12.216
t=0 0
m=audio 1024 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:0 PCMU/8000
```

NetSurveyorWeb™

- Multiple PacketScan™ probes can be used for network monitoring, with call detail reports exported to a central database
- Results can be accessed remotely using NetSurveyorWeb™, a simple web browser based application

The screenshot displays the NetSurveyorWeb web application interface. The top header shows the application name 'GL NetSurveyorWeb', a 'Refresh' button, and the 'Protocol Type' set to 'VOIP (SIP & H323)'. The user is logged in as 'GI'. The main navigation menu on the left includes sections for 'Quick CDR' (All Calls, Failed Calls, Passed Calls, Poor LMOS, Good LMOS, Longer Duration Calls, Voice Calls), 'Custom CDR' (CDR), 'Failed' (Failed), 'Default KPIs' (Basic KPIs), 'MailBox', 'Config', 'Admin', and 'Utilization'. The main content area is titled 'Quick CDR \ All Calls' and shows a date range of '2018-07-05' to '2018-07-05' with a time range from '00:00:00' to '23:59:59'. The 'Quick Search' field contains 'Trafficsumid'. The table below lists call records with columns for SI No, Calling Number, Called Number, Starttime, Duration, Call Success, Failure Cause, Listening Mos1, Listening Mos2, Payload1, and Page. The table contains 12 rows of call data.

SI No	Calling Number	Called Number	Starttime	Duration	Call Success	Failure Cause	Listening Mos1	Listening Mos2	Payload1	Page
1	0159@192.168.12.163	0159@192.168.12.164	2018-07-05 12:12:47.134	00:01:00.024	1	0	3.02	3.02	SPEEX/8000	19
2	0160@192.168.12.163	0160@192.168.12.164	2018-07-05 12:12:47.134	00:01:00.024	1	0	3.02	3.02	SPEEX/8000	19
3	0161@192.168.12.163	0161@192.168.12.164	2018-07-05 12:12:47.134	00:01:00.024	1	0	4.16	4.16	SPEEX/8000	19
4	0158@192.168.12.163	0158@192.168.12.164	2018-07-05 12:12:47.104	00:01:00.024	1	0	4.16	4.16	SPEEX/8000	19
5	0157@192.168.12.163	0157@192.168.12.164	2018-07-05 12:12:47.094	00:01:00.024	1	0	4.16	4.16	SPEEX/8000	19
6	0156@192.168.12.163	0156@192.168.12.164	2018-07-05 12:12:47.094	00:01:00.024	1	0	3.02	3.02	SPEEX/8000	19
7	0155@192.168.12.163	0155@192.168.12.164	2018-07-05 12:12:47.064	00:01:00.024	1	0	4.16	4.16	SPEEX/8000	19
8	0153@192.168.12.163	0153@192.168.12.164	2018-07-05 12:12:47.044	00:01:00.024	1	0	4.01	4.01	iLBC_15_2/8000	19
9	0154@192.168.12.163	0154@192.168.12.164	2018-07-05 12:12:47.044	00:01:00.024	1	0	3.95	3.95	iLBC_13_33/8000	19
10	0152@192.168.12.163	0152@192.168.12.164	2018-07-05 12:12:47.034	00:01:00.024	1	0	3.98	3.98	EVR CB/8000	19
11	0151@192.168.12.163	0151@192.168.12.164	2018-07-05 12:12:47.024	00:01:00.024	1	0	3.98	3.98	EVR CB/8000	19
12	0150@192.168.12.163	0150@192.168.12.164	2018-07-05 12:12:47.014	00:01:00.024	1	0	3.77	3.77	EVR CB/8000	20

Thank you